Invicta DAC Owner's Manual

Resonessence Labs

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Introduction

Music means so much to the human soul. The sounds and songs that accompany us through our lives reflect the emotion of forgotten times, re-inspire us to dream as we did before, hasten our heart beat, and flood over us with rekindled emotions. We were ready for the music revolution: transistor radios of the 1960's began the change, and through cassette players to portable CDs and to the early MP3 players, we reveled in the music of the late 20th century. The digital music revolution unfolded and showed us how to create, deliver and experience a choice of music unimaginable even a few years before. We are immersed in music: in our cars, on our phones, in our ears as we walk the streets. We owe this to the fabulous combination of digital and analog electronics: digital to flawlessly transport and store our music, yet still and always, analog to render it to our ear. We all know what the three letters "DAC" mean - Digital to Analog Converter - the key element in the digital music device and a part of the revolution.

The Invicta is an Audio DAC at a performance level that makes it a milestone in the evolution of digital audio.

Up until this point in time there has always been a trade-off in digital audio - the technology was not quite up to it, and commercial realities intruded. Costs had to be appropriate for the mass market. CDs squeezed as much data onto the disk as they could, sampling at 44.1khz was barely mathematically justified, but the digital was difficult and expensive, better to use a steep analog filter than sample faster. And the choice of 16bit encoding; who could dispute that 16 bits of resolution was not more than enough to beat the performance of the vinyl records? Early MP3 players used an admirably clever encoding scheme that put a tempting number of tracks on a relatively small FLASH memory card. This enabled a generation of profitable products, but few would argue that MP3 was anything more than an acoustic illusion, burying detail of the music behind louder sounds in the assumption that the human ear would not miss it.

The Invicta DAC plays uncompressed music in 24bit format up to 192kS/s. This far exceeds the performance levels of MP3.

As the 21st century begins, the technology of digital music leaps forward. It is no longer necessary to compromise the artist by compressing her music. The older trade-offs are no longer relevant. We can embed unadulterated perfection into the digital domain and, at a time of our choosing, recreate every essence of the living sound. You will never go back; hearing 24 bit music sampled at 192khz preserved and rendered bit-perfect without compression is an unforgettable experience. The mind builds an audio picture so intense in detail, so clearly spatially resolved, that the listener is inclined to disbelieve the eyes in favor of the ears that are telling him he is at a different place and time. The audio reality exceeds that of the vision, and the listener is transported into the time and place of the recorded performance.

A rational reader should be skeptical of such claims of great benefit from yet more technological power because, many times, the art of audio engineering has rested more upon the engineering than the art. To even suggest that the ear can benefit from resolutions higher than 16 bits is controversial: what experiment supports the assertion that 24 bits are perceived as any improvement on 16 bits? Why ever exceed 20khz of bandwidth when we know that we cannot hear a 20khz signal? And does not the rigor of mathematical analysis teach that 44.1khz sampling can indeed preserve 20khz - so where does any further technological benefit lie? What shall we conclude about the obfuscation that sigma-delta ($\Sigma\Delta$) modulators apply to the signal - are they not just another MP3-like slight-of-hand designed to reduce the manufacturer's costs? Indeed it is true that in the race to exploit the exploding market for personal audio devices, rationalizations such as these (no need for more than 16bits, no need for high sampling rates, use of low cost $\Sigma\Delta$ based DACs etc) have driven mass-market designs.

Careful science shows that the ear is a remarkable and subtle instrument. Immeasurable differences can be perceived by skillful listeners.

To excel in audio engineering and design is a challenging task. It is challenging because the reality of it differs from that which is suggested by a simple analysis. The reality is that individuals with audiophile credentials (so called "golden ears") do exist, and one can be trained to listen critically to music and discern faults and imperfections that a casual listener does not hear. Some audio enthusiasts are outstanding in their ability to note differences that the engineer cannot find. Scientific double-blind tests confirm over and over again that two digital audio devices with identical specifications of noise, dynamic range, bandwidth and distortion can be reliably distinguished by the human ear. There is certifiably something that the ear perceives that our measurement

apparatus does not. Indeed, there are examples of outstanding audio designs where the skilled engineer has deliberately failed to optimize a certain parameter (the most common being signal to noise ratio) because listening tests consistently show that the lower quality design (as measured by the instrumentation) sounds better. All the high-end audio manufacturers employ staff trained to listen to new products and they only accept a design that passes a listening test. And the important point is this: these listening tests confirm that the high amplitude resolution (24 bits) and high sampling rate (96k or 192k) make a huge difference to the perception of the audio rendered by the digital device.

To understand Resonessence is to understand this: what if the most technically advanced device was also the one that produced the best results in listening tests? What if a device could absolutely and verifiably dominate in the technical measurements of signal to noise, dynamic range and distortion, jitter, in response flatness, in linearity of group delay, in out of band noise, in the subtleties of transient response and uniformity of noise in operating space, and yet also excel in listening tests? What if it could conform to expectations of modern interfaces, USB, SPDIF, word sync and add yet more such as SD Card based audio? The Invicta is this product.

The Resonessence Invicta DAC is technically superior, but more importantly, it sounds better because it is designed with feedback from audiophiles, not just from instrumentation measurements.

You are now in possession of the Resonessence Invicta DAC and you can confirm all the assertions above. Connect the Invicta to your quality amplifier and speakers, or use high quality headphones from the front panel, and listen for yourself. The Invicta is capable of rendering 24 bit audio at 192khz from all sources, the most convenient probably being the SD card. Relish your first experience of our product - we hope you find it enjoyable. This document is your guide. In the second section we describe how to use the Invicta in your system - all you need to know to get the best out of your purchase.

Refer to the first section for a quick start guide; this should be sufficient to show how to get the Resonessence Invicta running with either USB or the SD-Card.

The third section is the technical detail including everything measurable about the Invicta – those aspects that can be technically measured are reported here so that you can be assured you are in possession of a technically excellent product – the best we know how to build. You will note that we discuss unusual parameters over and above the industry

standard measurements. These are based on our experiments where we have discovered a correlation between what the audiophile trained listener perceives as good, and a measurable parameter or design aspect.

In section four of this user guide we share what we have learned as we have grown in the audio field. There are pitfalls and aspects of audio shared here that you may find helpful. For example, the use of Fourier analysis is ubiquitous for audio, but sometimes its faults and foibles are not sufficiently explained. Also, the Invicta uses the ESS Sabre32 DAC as the core element. This is a complex device that has many aspects of a $\Sigma\Delta$ modulator, but claims some difference. The manufacturer calls it a HyperStream modulator – you may be interested to know what that is and why it does not suffer from sigma-delta modulator problems. Finally, there are some so-called secrets of design in audio that are not really secrets – all the good manufactures know them – we think it may be helpful to explain them to our customers because we find that many audio enthusiasts who appreciate the Invicta product also appreciate the technical details.

Chapter 1

Quick Start Guide

Check the Mains Voltage The Invicta product is configured for the prevailing mains voltage at the delivery address, however as a precautionary measure, first confirm that the voltage setting on the power entry module on the back of the Invicta corresponds to the mains voltage in your region.

Take care to confirm that the voltage setting is correct for your mains supply.

Figure 1.1: The mains voltage is visible on the power entry module

The selected voltage is visible to the left of the power switch, in this picture 115V is selected. If the selected voltage is incorrect, refer to section 2.1

1.1 USB Audio to Headphones

1. Connect the power and USB interface cable on the rear panel, and connect the other end of the USB cable to the computer.

2. Turn on the power to the Invicta unit. After a short interval the computer should recognize the Invicta and it will be available in the system "Sound" dialog. Select this as the Default Device.

The Resonessence Invicta will be recognized by the Windows OS and be listed in the "Sounds" Dialog. Choose the Invicta as the Default Device.

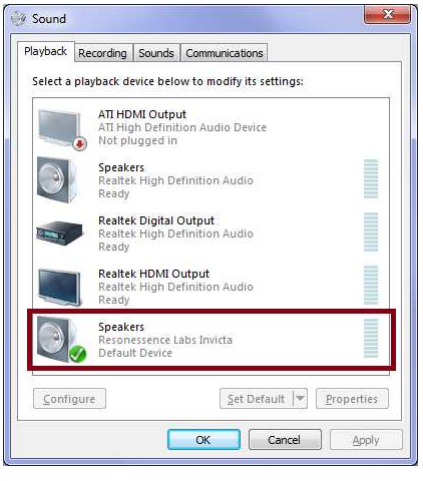

Figure 1.2: Resonessence Invicta is registered as a Speaker

3. Plug the headphones into either of the $\frac{1}{4}$'' headphone plugs A or B on the front panel and activate the headphone output with the button above the corresponding socket.

   When using two headphones the volume level may be different in each one. To correct this refer to section 2.5.

4. Press the "Source Select" button (right-hand of the four buttons below the display) until USB is the selected source. The red light

in the bottom left[1] will indicate that the system is waiting for USB data to be available from the computer. Any music now played on the computer is routed to the Invicta and played through the headphones.

1.2 SD-Card to Amplifier and Speakers

1. Connect the power cable on the rear panel and turn on the power.
2. Connect the XLR or RCA rear connectors to the Power Amplifier input ports on your audio amplifier.
3. Insert an SD-Card containing the source you wish to play: the display will show "SDHC Card Inserted"
4. The default configuration is to take data from the SD-Card. Rotate the multi-function knob to select the track you wish to play.
5. Press the multi-function knob and the track will start to play, press again to pause, rotate to change the volume level.

[1]That is, the USB Status lights in the bottom left, "USB Status" Figure 2.3

Chapter 2

Resonessence Invicta User Guide

2.1 Checking the Mains Voltage

The Invicta uses a linear power supply as opposed to a switching power supply[1] and consequently requires the user to set the expected mains voltage.

The Invicta does not contain an automatically adapting switch-mode power supply: they make too much noise. You must set the mains voltage by rotating the fuse holder.

Figure 2.1: The mains voltage is visible on the power entry module

The selected mains voltage is visible on the power entry module (PEM) at the rear of the unit. The Invicta is configured for the mains voltage

[1] Refer to section 4.4 for more details

of the shipping address, but if the system is shipped elsewhere or if the mains voltage needs to be changed, then carefully open the fuse door, extract and re-insert the fuse holder appropriately to select the new voltage.

Choose 230V or 115V - whichever is closest to your mains voltage.

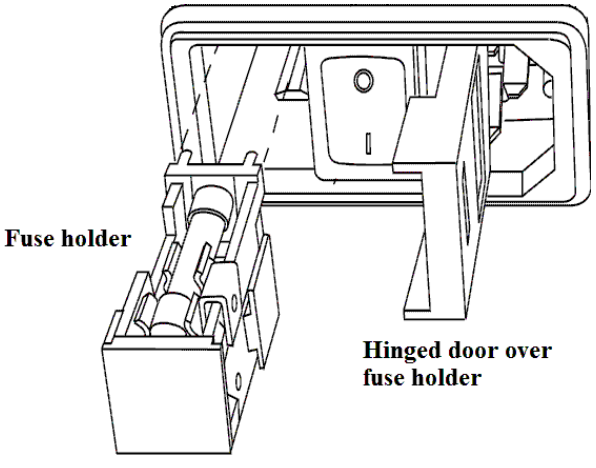

Figure 2.2: Drawing showing the fuse door and fuse holder

Close the door and verify that the expected voltage (230V or 115V) is visible in the window.

2.2 A Note on Output Levels

The output amplitudes of the Invicta are designed to drive typical power amplifier input levels, not pre-amplifier input levels.

The Invicta output levels are designed to drive a power amplifier input and are therefore higher than commonly used for the Analog Inputs of audio equipment that combines pre-amplification with a power amplifier. Take care not to overdrive your audio equipment inputs. Ideally, use the power amplifier input on your equipment, not the auxiliary analog inputs.

The XLR differential output amplitude for 0dB is $5V$rms and the RCA output amplitude for 0dB is $2.5V$rms. This is typically higher than the levels expected at the Analog Input of many home audio devices because the Invicta is designed to drive a power amplifier input.

The Invicta at power-up presets the volume level to $-30dB$.

2.3 Front Panel Controls

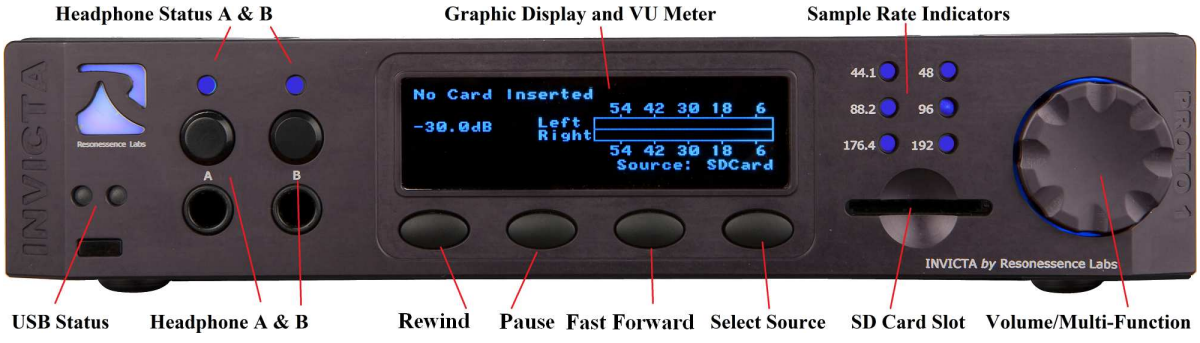

Figure 2.3: Invicta front panel controls

2.4 Rear Panel Connectors

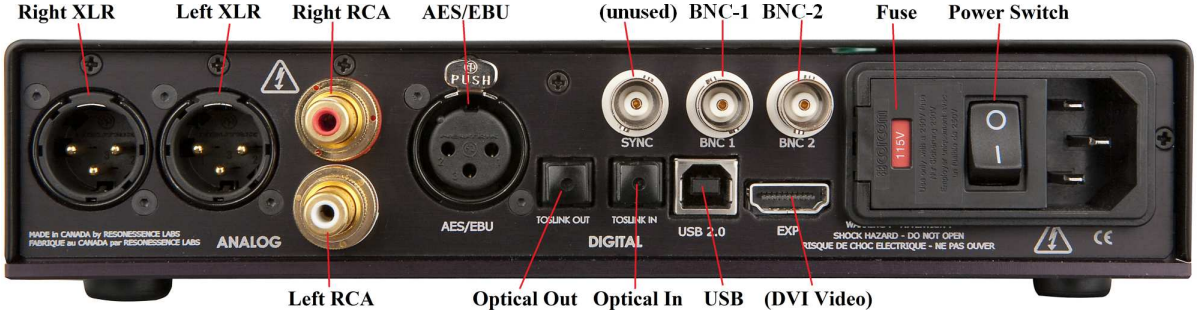

Figure 2.4: Invicta rear panel connections

2.5 Calibrating the Headphone Levels

Invicta has two headphone output jacks to enable two sets of headphones to be used simultaneously. The Volume Control dial will adjust the volume to both the left and right headphone jacks when turned. The left and right headphone control buttons allow you to calibrate the volume control for each pair of headsets. If the volume is too low or two high for one set of headsets, hold down the headphone control button above the headphone jack and adjust via the Volume Control dial and then release.

Headphones can have different sensitivities. Follow the procedure here to set the levels to be the same so that two people can listen at the same time.

2.6 Digital Data Sources

Invicta processes digital input data from one source and sends the analog reconstruction to all outputs at the same time.

The Invicta plays from one data source at a time, the selected data source is sent to each output simultaneously. Therefore, you can listen via the headphones or via the RCA or XLR outputs at the same time.

2.6.1 SD Card

The SD-Card reader will read files from a Class-10 SD-Card in WAV file format. The file can be of up to 192kS/s and 24bit data width.

Insert the SD Card and press the "Source Select" button (right most of the four buttons) until the SD-Card source is selected. The selected source is visible in the bottom right of the display area. Rotate the multi-function knob to select the track name you wish to play – as the knob rotates, the track is visible in the upper left area of the display.

Playback from the SD-Card maybe Paused, Rewound and Fast Forwarded from the front panel buttons.

Press the multi-function knob to start playing and then adjust the volume with the multi-function knob.

To stop playback, press the multi-function knob again.

Invicta will play each track on the SD Card in sequence until the end of the last track on the card.

To pause a track, press the pause button. To re-start, press the pause button a second time or press the center of the Volume/Multi-Functional dial.

You can rewind gradually through a track by holding down the Rewind button. The track will restart automatically as soon as the button is released. To quickly rewind to the beginning of a track, press and release the rewind button once. To rewind to the beginning of the sequence of tracks on the SD Card, press the rewind button twice in quick succession.

You can Fast-Forward gradually through a track by holding down the Fast-Forward button. The track will restart automatically as soon as the button is released. To quickly Fast-Forward to the end of a track, press and release the Fast-Forward button. To Fast-Forward to the end of the sequence of tracks on the SD Card, press the rewind button twice in quick succession

2.6.2 Digital Data Inputs

The Invicta accepts digital data in the Sony/Philips Digital Interface Format (S/PDIF) from three different sources. Each data source may transmit up to $192kS/s$ data at up to 24 bits. The Invicta recognizes the data rate, bit width etc. and configures appropriately without any user interaction. The data rate being processed is indicated on the front panel LEDs.

Invicta accepts only uncompressed stereo data and mutes in the presence of compressed data. De-emphasis following the RIAA specification is applied if the pre-emphasis bit is set in the S/PDIF data stream.

Digital data sources are galvanically isolated via transformers (or inherently as in the case of the TOSLINK) to prevent unexpected ground or other current flow which can create hum[2].

The Invicta has the TOSLINK Optical connection that is probably the easiest to use in a consumer application. The BNC and AES/EBU inputs conform to professional expectations of format and electrical isolation and are less common in consumer equipment.

If an RCA style SPDIF connector is a necessity, use a commercially available BNC to RCA converter on one of the BNC ports.

- **BNC (AES3id) S/PDIF Inputs**

  Two selectable 75Ω BNC connectors are transformer coupled and internally terminated for connection to a 75Ω co-axial (single ended) cable. Select either of the BNC data sources by pressing the right most "Source Select" button until "BNC-1" or "BNC-2", as appropriate, is visible in the display window[3].

  The BNC S/PDIF in a managed 75Ω environment is optimum for relatively long distance (significantly greater than one hundred meters) digital data transmission.

  The outer shells of the BNC connectors are isolated: they float relative to the chassis potential in order to prevent ground currents and preserve the galvanic isolation.

- **AES/EBU S/PDIF Input**

  An AES/EBU (Audio Engineering Society / European Broadcast Union) female XLR connector is transformer coupled and internally terminated for connection to a nominally 110Ω differential shielded twisted-pair (STP) cable. Select the AES/EBU data source by pressing the right most "Source Select" button until "AES/EB" is visible in the display window.

[2]Note that the screen of the AES/EBU digital data cable is at chassis potential.

[3]The third BNC input labeled "Sync" on the back panel is currently unused and is provided in anticipation of a future firmware update.

Although differential and conveniently sharing the wiring of analog XLR style connectors, impedance levels and losses of the AES/EBU digital connection standard limit its application to distances of less than a few tens of meters.

The screen terminal of the AES/EBU XLR is at chassis potential: the twisted pair within the cable is galvanically isolated from this screen by the internal transformer.

- **TOSLINK Optical S/PDIF Input** TOSHiba-LINK is an optical connection as defined by Toshiba Corporation. The Invicta expects the optical fiber to carry the S/PDIF data format (not PCM). TOSLINK is provided primarily for consumer applications, but similarly to the BNC and AES/EBU, Invicta will accept the same framing and bit standards (responding to the pre-emphasis and compressed bits) as on the other digital data sources. Select the TOSLINK data source by pressing the right most "Source Select" button until "TOSX" is visible in the display window.

2.6.3 USB

For a "Quick start" to using the USB see section 1.1.

The USB (Universal Serial Bus) interface is perhaps the most popular, and certainly the most requested, interface format. The convenience of integration into the personal computer, access to on-line content and the familiarity of the unified Windows or Mac user interface make it an appealing and practical means to access digital music.

Don't forget to select USB as the signal source with the "Source Select" button.

The Invicta may be connected via the standard Type-B connector on the back panel to a USB-2[4] port on the computer. Upon connection, and when powered on, the operating system will recognize Invicta as an audio device. Selecting the Invicta as the output device allows any player (such as Windows Media Player etc.) to be used, and tracks selected within that application will be streamed via USB to the unit. Note that the volume control still operates on the data, and the VU meter shows the output signal level.

Select the USB as the data source by pressing the right most "Source Select" button until "USB" is visible in the display window.

[4]Invicta expects a "Hi-Speed" USB port, that is a USB 2.0 port.

In USB mode, the Invicta master clock is the timing source and data from the computer is requested and buffered to maintain synchronization. Clearly, if the player application on the computer ceases to send data, Invicta will stop playing and the USB underflow light (bottom left of the front panel) will glow red to indicate this condition. Also, depending on the detail of the player application, a request to pause in the application may result in the repetition of the last sample point, a series of zero values, or no data at all being present on the USB bus. If no data is present, the buffer will underflow and the indicator light illuminate, but if the other conditions of repeated value or series of zero values are detected then the VU meter is adjusted to show no signal.

2.7 Updating Firmware

All elements of the Invicta are programmable. From time to time Resonessence may upgrade features by providing new firmware modules.

There are three separate firmware modules in the Invicta:

- The firmware that controls the USB Interface (referred to as the "USB Firmware" or "USB Fw").
- The code that runs within the configurable hardware engine (referred to as the "Micro-Controller Code" or "MCU Fw")
- The configuration code that creates the configurable hardware engine in the Invicta (referred to as the "FPGA Code" or "FPGA")

To update any or all of these software modules:

1. Connect the power cord to the unit and a USB cable to the MSWindows computer. Turn on the unit – within a few seconds Windows will recognize the device
2. *Before updating, verify that the Invicta is not being used by any application. For example, Skype may have found and used the Invicta as the sound output device. If any program is using the Invicta, update may fail. To be sure, temporarily select another output device as the default Windows sound device.*
3. Double click on the provided **dfu.exe** application. The following dialog box will open:

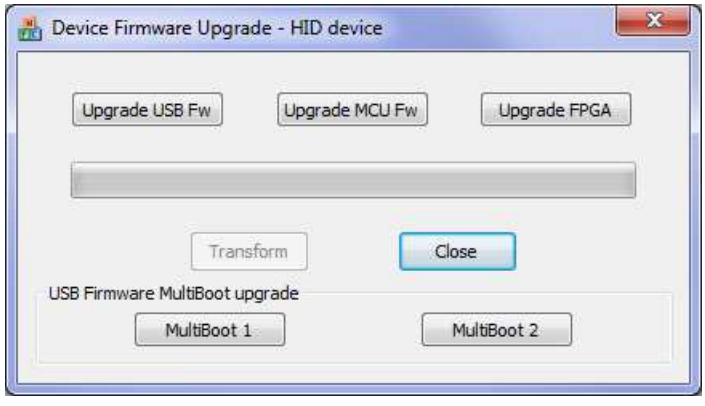

Figure 2.5: The Firmware Upgrade Utility Dialog Box

4. If you wish to update the USB Interface Firmware, click the button named "Update USB Fw" and in the file dialog box choose the appropriate file. The file has the type "iic".

The USB interface (a silicon chip from Cypress Semiconductor) can be upgraded if necessary.

5. After a few seconds, the USB firmware will be updated. The internal device handling the USB is from Cypres Semiconductor, and a reminder to cycle the power will appear:

Wait for Cypress Reset
Cypress firmware code is downloaded. You have to reset or power cycle the board to continue.
Device Firmware Upgrade - HID device
Upgrade USB Fw
Upgrade MCU Fw
Upgrade FPGA
Transform
Close
USB Firmware MultiBoot upgrade
MultiBoot 1
MultiBoot 2

Figure 2.6: USB Firmware Upgrade Completed

6. Turn off the power to the Invicta, wait a few seconds for Windows to un-register the device, then turn on the Invicta again.

7. If you wish to update the Micro Controller Code, click the button named “Update MCU Fw” and in the file dialog box choose the appropriate file. Often the name “MCU” will be part of the file name. The file has the type “bin”.

8. After a few seconds, the MCU firmware will be updated. The following dialog appears:

The MCU is the main controller of the Invicta. It is micro-code running within a high performance Field Programmable Gate Array (FPGA).

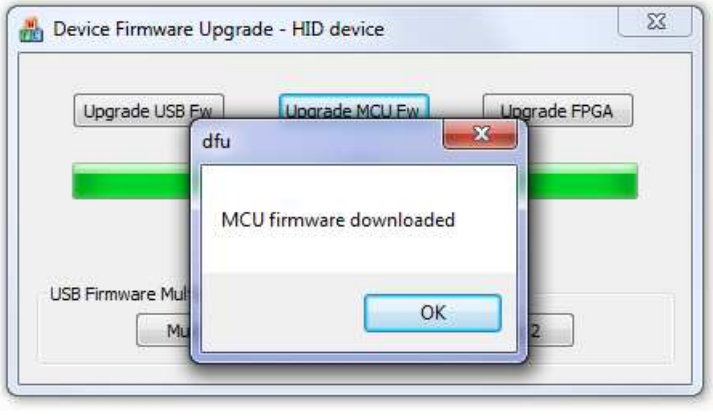

Figure 2.7: MCU Firmware Download Completed

9. Click “OK” and turn off the power to the Invicta once more, wait a few seconds for Windows to un-register the device, then turn on the Invicta again.

10. If you wish to update the FPGA Code, click the button named “Update FPGA” and in the file dialog box choose the appropriate file. Often the name “FPGA” will be part of the file name. The file has the type “bin”.

The FPGA itself may be upgraded to add extra features and operation codes to the MCU.

11. After a few seconds the FPGA Code will be updated. The following dialog appears:

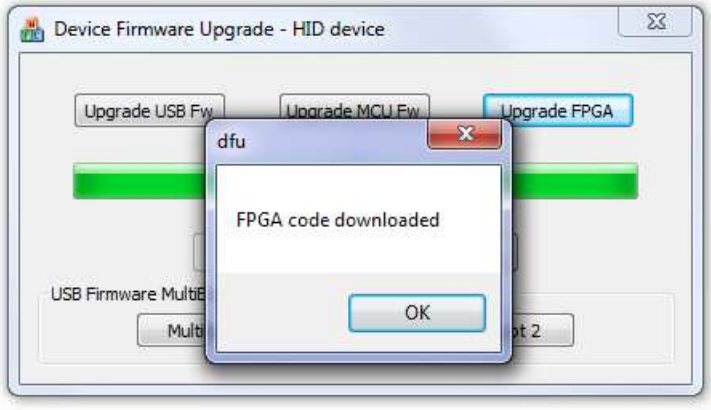

Figure 2.8: FPGA Code Download Completed

12. Click “OK” and turn off the power to the Invicta once more, wait a few seconds for Windows to un-register the device, then turn on the Invicta again.

Chapter 3

Technical Measurements

All measurements are taken with the Audio Precision Model 2722. Below is a typical graph produced by the instrument with notation of the elements of the graph. Many of the graphs are in the frequency domain, that is, they are Fourier Transforms (sometimes referred to as Fast Fourier Transforms or FFT's). Refer to section 4.3 for details on the FFT.

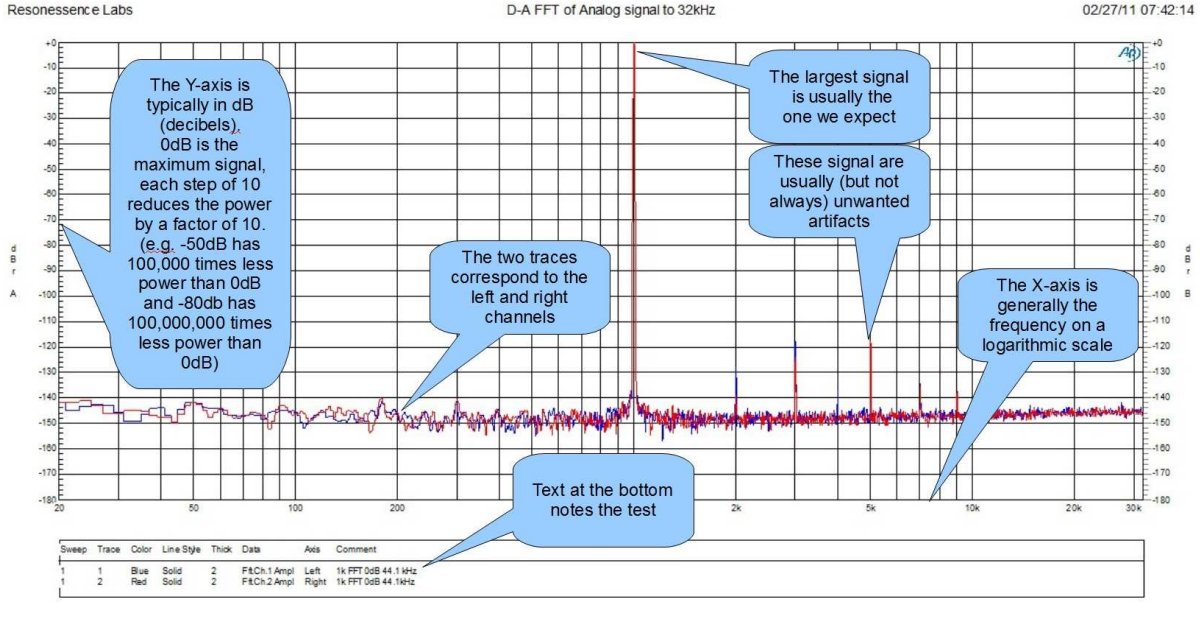
Resonessence Labs
D-A FFT of Analog signal to 32kHz
02/27/11 07:42:14
The Y-axis is typically in dB (decibels). 0dB is the maximum signal, each step of 10 reduces the power by a factor of 10. (e.g. -50dB has 100,000 times less power than 0dB and -80db has 100,000,000 times less power than 0dB)
The largest signal is usually the one we expect
These signal are usually (but not always) unwanted artifacts
The two traces correspond to the left and right channels
The X-axis is generally the frequency on a logarithmic scale
Text at the bottom notes the test
Sweep Trace Color Line Style Thick Data Axis Comment
1 1 Blue Solid 2 FftCh.1 Ampl Left 1k FFT 0dB 44.1 kHz
1 2 Red Solid 2 FftCh.2 Ampl Right 1k FFT 0dB 44.1kHz

Figure 3.1: Annotated example of a typical AP2722 graph

3.1 20Hz FFT 0dB 44.1kS/s

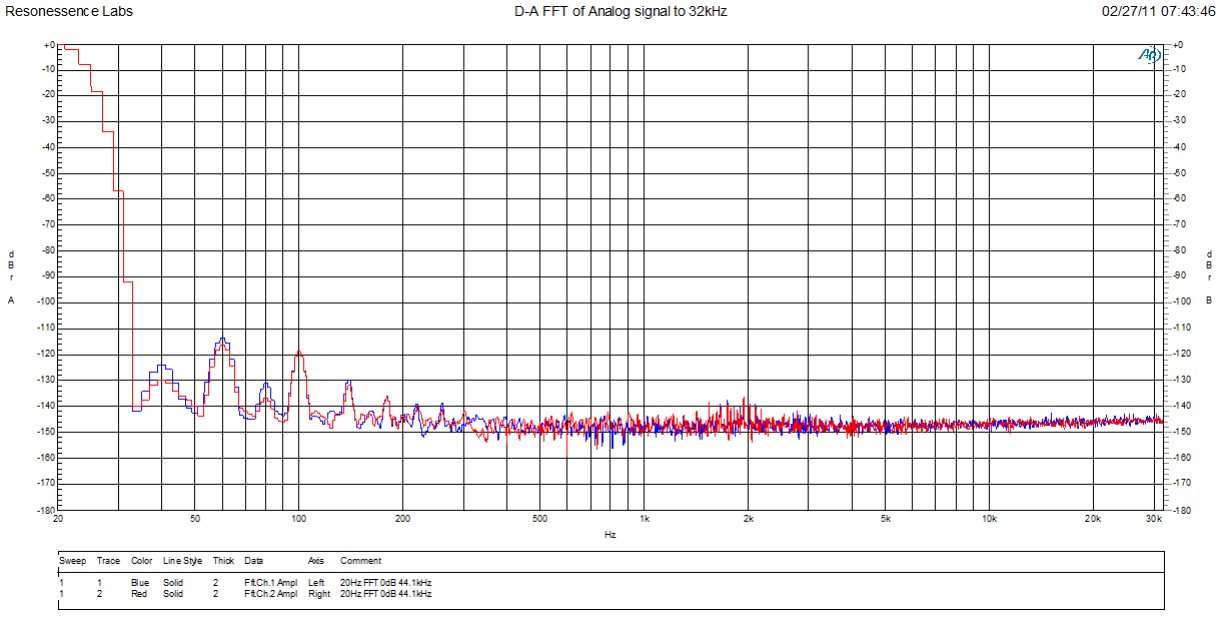

Figure 3.2: $20Hz$ Signal Sampled a4 $44.1kHz$ (as from a CD)

The Invicta shows low harmonic distortion in the $20Hz$ test. No high harmonic terms are present and no evidence of spurious tones.

This FFT of a $20Hz$ signal shows no roll-off for low frequencies and very low distortion up to high harmonics. Testing near the low frequency end of the frequency range can also expose spurious tones (often called idle tones) that some forms of $\Sigma\Delta$ modulators are susceptible to.

Refer to the $1kHz$ result on the next page for more details of the FFT based tests.

3.2 1kHz FFT 0dB 44.1kS/s

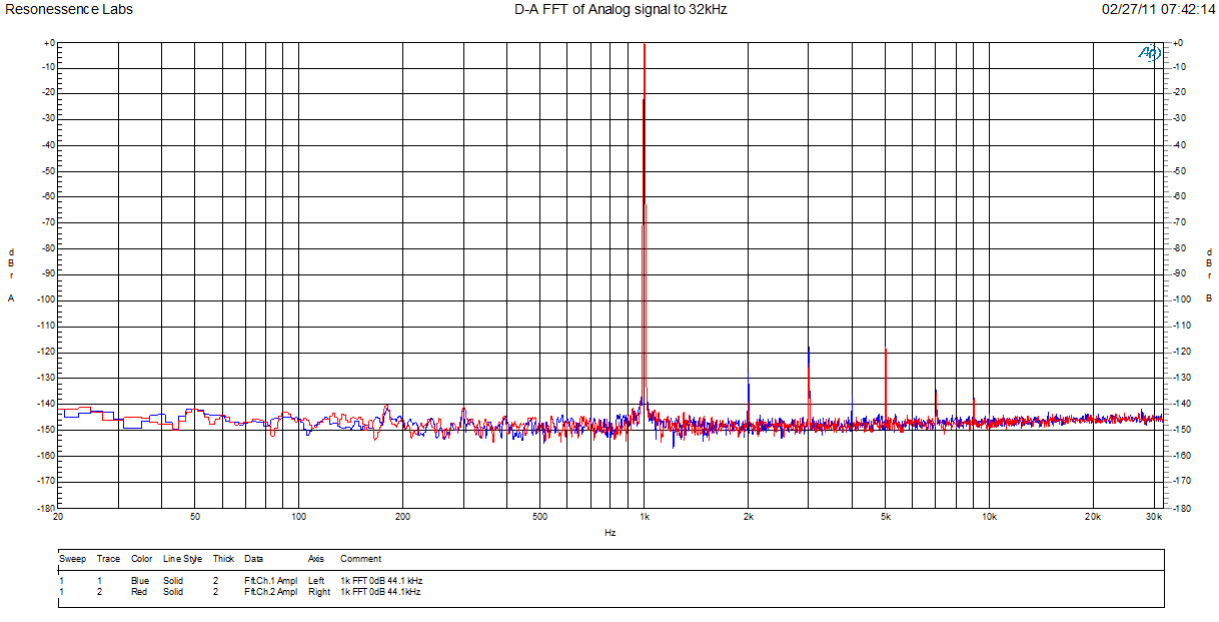

Figure 3.3: $1kHz$ Signal Sampled at $44.1kS/s$ (as from a CD)

The FFT of a $1kHz$ signal is the most commonly used technical measurement. This is the Invicta creating a single tone at $1kHz$ from digital data delivered at a rate of $44.1kS/s$ which is the rate encoded on a CD. The main component measured by the AP2722 is the $0dB$ signal at $1kHz$ – everything else is an artifact of the reproduction. The first thing we observe is that there is a "noise floor" below $-140dB$ and there are "spikes" sticking out of that noise floor to about $-120dB$. Close inspection shows that the highest ones are $3kHz$ and $5kHz$ – these are the 3rd and 5th harmonic of the $1kHz$ applied. "Harmonic" is the term used to describe simple multiples of the main signal and harmonics are desirable characteristics of musical instruments: they give the musical instrument its characteristic sound, as from a Clarinet or Oboe. However, in an electronic device, harmonics of the main signal are unwanted: they represent failures on the part of the electronic device to create just the main signal provided in the digital data stream. The electronic device is sometimes specified by the Total Harmonic Distortion (THD) which is the ratio of the main signal to the sum of all the harmonic terms, but more commonly the specification of Total Harmonic Distortion plus Noise (THD+N) because this can more easily be measured.

The FFT at $1kHz$ shown here demonstrates that the Invicta has a THD+N of $-114dB$.

3.3 7kHz FFT 0dB 44.1kS/s

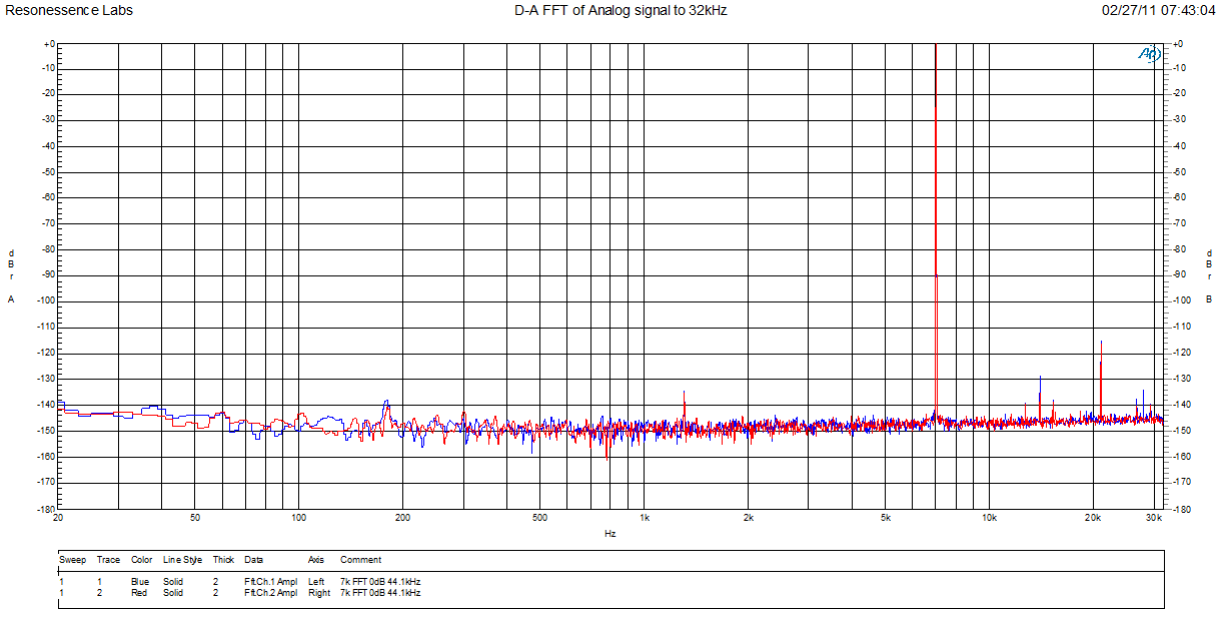

Figure 3.4: $7kHz$ Signal Sampled at $44.1kS/s$ (as from a CD)

The harmonics are remaining at the same very low amplitude for each frequency tested. The Invicta has no frequency dependent distortion.

The FFT of a $7kHz$ signal will have a third harmonic component close to the upper end of the band when a $44.1kS/s$ data rate is used. That harmonic is just visible here; it remains of low amplitude and nothing anomalous is occurring. The fifth harmonic is not present and no evidence of it "reflecting back" below the $20kHz$ bandwidth is present.

Note that the third harmonic of the $7kHz$ tone is not increased in amplitude – it is the same amplitude as was present in $20Hz$ and $1kHz$ tests. This demonstrates that there is no hint of non-linear limiting (slew rate or similar) in the DAC.

No "rumble" – low frequency artifacts – are visible in the frequencies below $7kHz$.

3.4 20kHz FFT 0dB 44.1kS/s

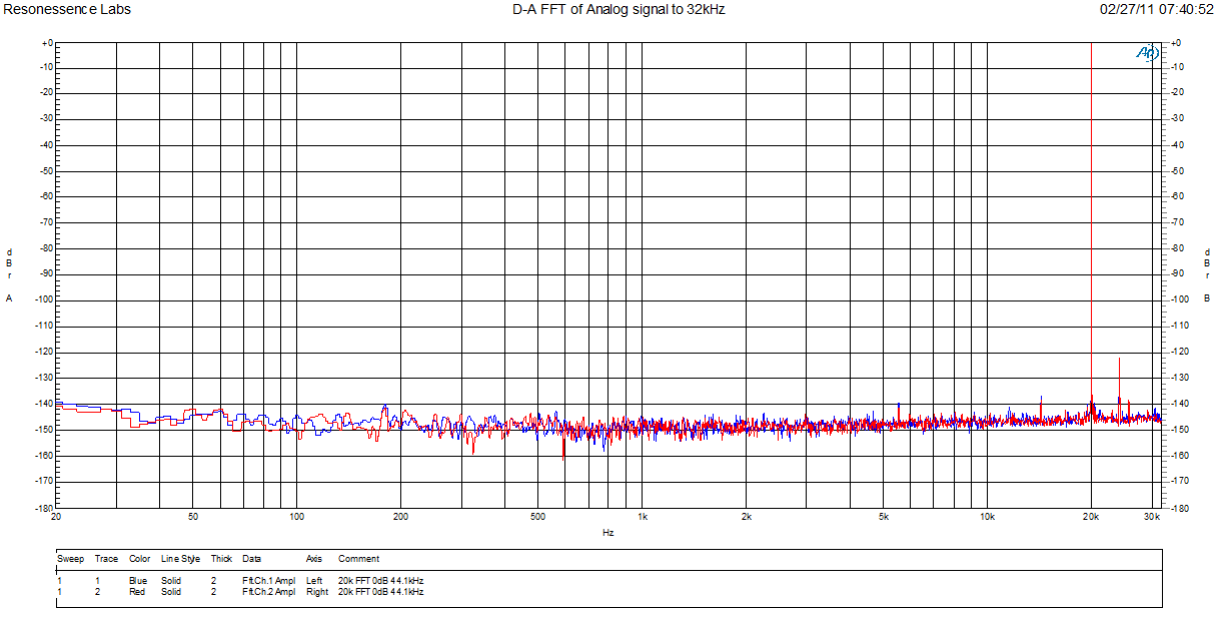

Figure 3.5: $20kHz$ Signal Sampled at $44.1kS/s$ (as from a CD)

Applying a $20kHz$ signal in the $44.1kS/s$ data rate stresses the band limiting digital filter (sometimes called the "Anti-Aliasing" filter). The $44.1kHz$ data rate is a legacy of the CD era and is just about as low a sampling rate as can be used while retaining a $20kHz$ bandwidth.

The digital anti-aliasing filter is explored in this FFT: it is working very well, suppressing the alias tone to less than $< -120dB$.

Because the sample rate is so low, the data stream carrying a $20kHz$ signal must also be carrying a $44.1kHz - 20kHz = 24.1kHz$ signal as well. This is a mathematical consequence of the low sample rate and is an "image" of the $20kHz$ signal. The task of the digital band limiting filter is to remove this artifact and render the signal into a substantially higher sample rate domain for further processing. The performance of the digital band limiting filter can be assessed by looking for the $24.1kHz$ artifact. It is a very low: $< -120dB$.

3.5 1kHz FFT -90dB 44.1kS/s at 16bits

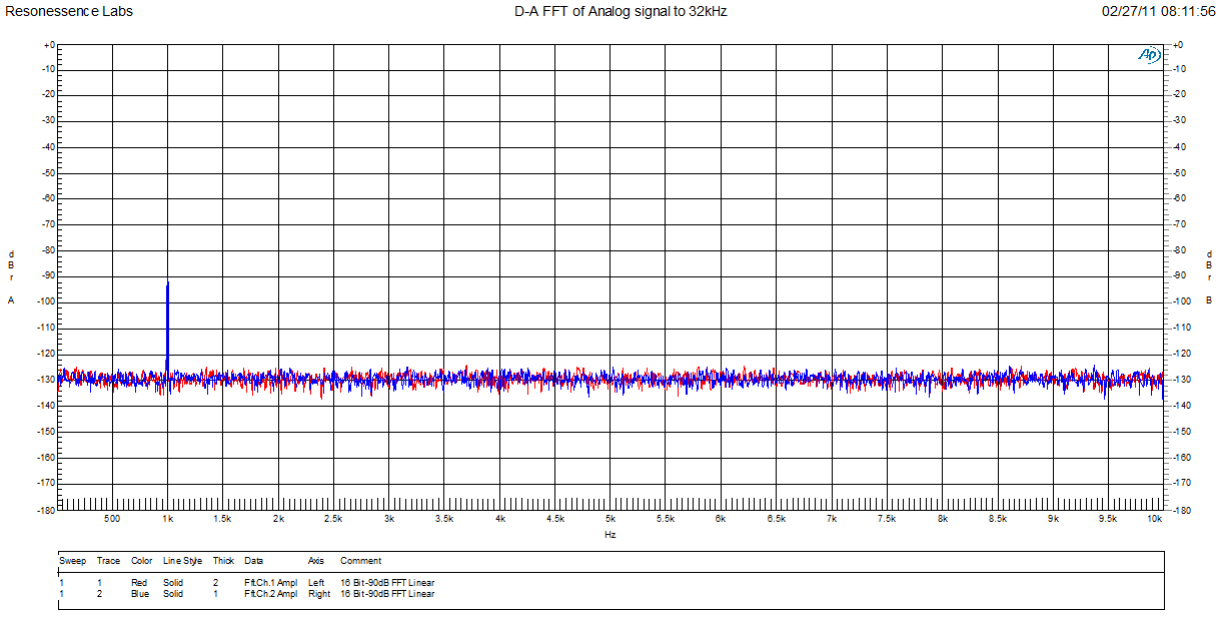
Resonessence Labs
D-A FFT of Analog signal to 32kHz
02/27/11 08:11:56
dBrA
dBrB
Hz
Sweep Trace Color Line Style Thick Data Axis Comment
1 1 Red Solid 2 FftCh.1 Ampl Left 16 Bit-90dB FFT Linear
1 2 Blue Solid 1 FftCh.2 Ampl Right 16 Bit-90dB FFT Linear

Figure 3.6: Low Level $1kHz$ Signal Sampled at $44.1kS/s$ 16bits

The noise floor is flat and entirely due to quantization noise in the incoming data. No idle tones are present.

If the amplitude of the main tone is reduced we can expect the harmonics to disappear into the noise floor. This test applies a small signal ($-90dB$) to the DAC and verifies that all the harmonics are gone.

In this situation the noise floor is determined by the larger of the DAC inherent noise and the quantization noise of the incoming data stream. Since this is a 16 bit data stream the quantization noise far exceeds any inherent DAC noise: this results in a flat and undisturbed noise floor at the $-130dB$ level and this is entirely due to the use of 16 bit data.

Certain forms of Noise Shaping loop (often simply called $\Sigma\Delta$ modulators) suffer from anomalous patterns that persist in the loop. These are called "idle tones" or "limit cycles" and are self-sustained perturbations in the almost chaotic state variables that do not die away with time.[1] No idle tones are visible in this data.

[1] To some degree, idle tones correspond to annoying "rattles" that your car may make but only when going at certain speeds. It is important to put the DAC into a state that encourages these idle tones to verify they are not present.

3.6 1kHz FFT -90dB 44.1kS/s at 24bits

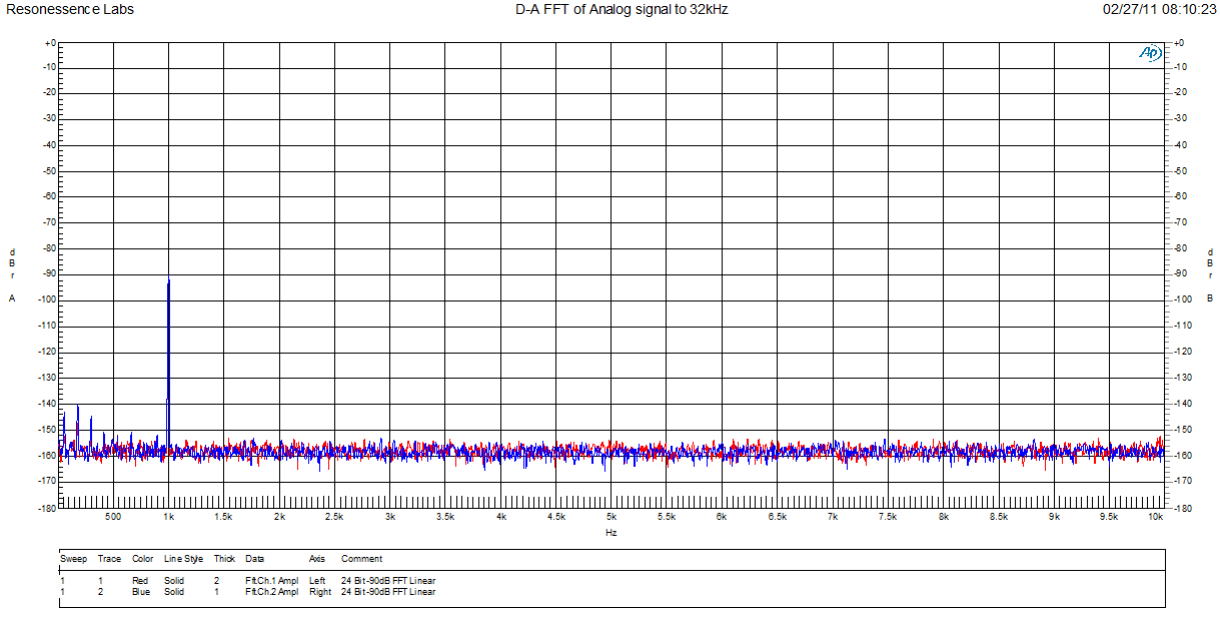
Resonessence Labs
D-A FFT of Analog signal to 32kHz
02/27/11 08:10:23
dBr A
dBr B
Hz
Sweep Trace Color Line Style Thick Data Axis Comment
1 1 Red Solid 2 FftCh.1 Ampl Left 24 Bit -90dB FFT Linear
1 2 Blue Solid 1 FftCh.2 Ampl Right 24 Bit -90dB FFT Linear

Figure 3.7: Low Level $1kHz$ Signal Sampled at $44.1kS/s$ 24bits

As in the 16 bit test on the previous page, there is no evidence of low-level harmonic distortion in this $-90dB$ test. However, the data stream has about $46dB$ lower quantization noise and is now below the level of the DAC. Consequently, the noise floor in this graph is entirely due to the inherent noise in the DAC itself. The noise floor is now about $-155dB$ and any very low level idle tones or other anomalies would be visible. There are none.

The DAC noise floor is visible in a low-level FFT with 24bits. It is free from idle tones and spurious responses.

This is in large part due to the ESS Sabre DAC used in the Invicta. It has a noise shaping loop which differs in detail from a conventional $\Sigma\Delta$ modulator (the manufacturer calls it a "HyperStream" modulator).[2]

The signals at the $-140dB$ level below $500Hz$ are harmonics of the mains cycle ($60Hz$ in this case) getting into the signal path probably through electro-magnetic induction. $-140dB$ is an exceptionally low level of power line interference.

[2]The HyperStream modulator adds a third constraint to the noise shaping loop. It and the $\Sigma\Delta$ modulator must quantize the signal in amplitude and time while shaping the noise, but the HyperStream also constrains the loop dynamics to fix the feedback frequency. This makes it virtually impossible for self-sustained low level oscillations (idle tones) to persist in the loop.

3.7 1kHz FFT -140dB 44.1kS/s

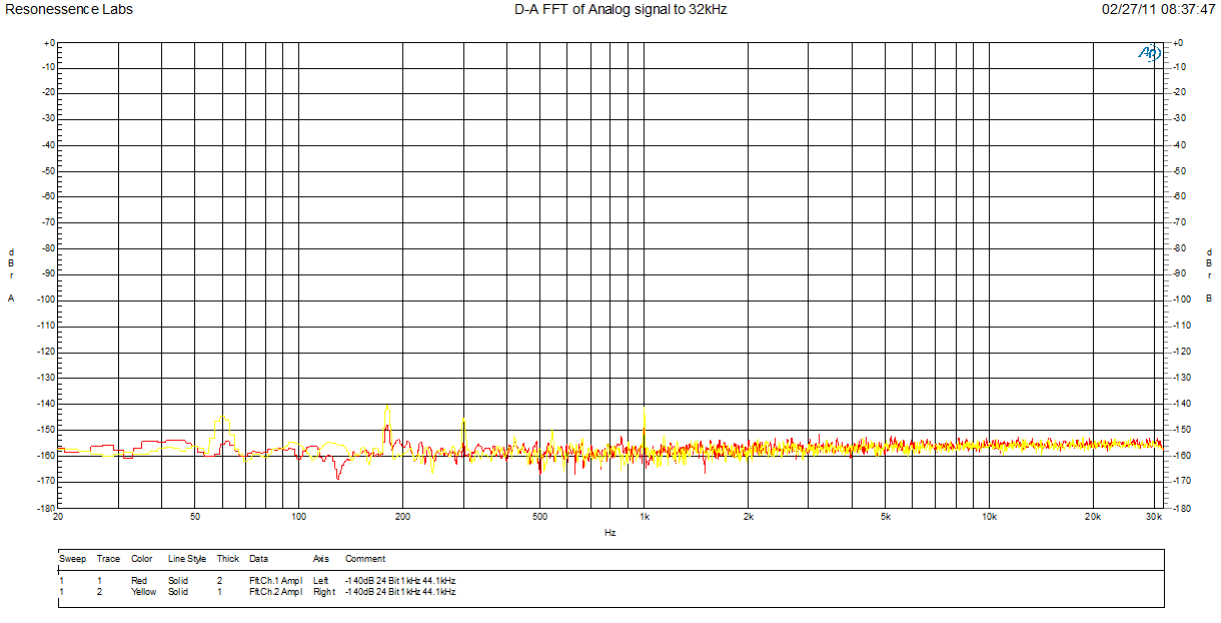

Figure 3.8: Very Low Level $1kHz$ Signal Sampled at $44.1kS/s$ 24bits

The Invicta has a very low noise floor: so low that a signal of only $0.2\mu V$ can be clearly seen in the output.

This FFT is approaching the limit of the instrumentation used to measure the Invicta. Here, a very low level $-140dB$ signal is applied in the 24 bit data stream. This generates a signal of only $0.2\mu Vrms$ at the output[3]. Note that once again there is no evidence of idle tones or spurious response.

Close examination of this graph supports the assertion on the previous page: the $-140dB$ tones in the sub $500Hz$ region are now clearly seen to be power line induced noise. Due to the logarithmic X-axis used in this plot, they can be confirmed to be 60, 180 and $300Hz$ – harmonics of the Canadian power line frequency ($60Hz$) where this Invicta was measured.

[3]It is a testament to the engineering expertise in the AP2722 that it can resolve such a low level signal.

3.8 1kHz FFT 0dB 44.1kS/s Headphone

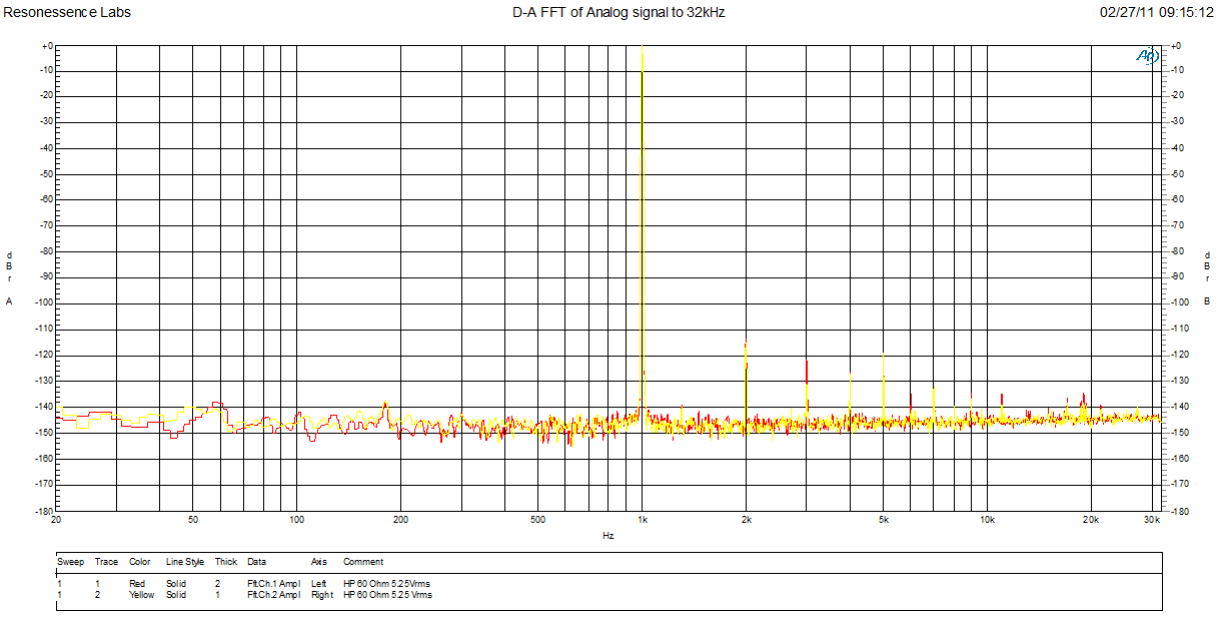

Figure 3.9: Head Phone Output

The Invicta has two types of analog output: the balanced XLR outputs on the rear (with subsidiary unbalanced RCA outputs for those applications that require a single ended output) and a pair of standard $\frac{1}{4}$'' TRS sockets ("Headphone Jack" sockets) for convenient connection to high quality headphones on the front panel.

In addition to the high performance XLR outputs on the rear, the Invicta provides dual headphone outputs on the front panel.

The headphone sockets are slightly inferior in terms of performance to the XLR outputs, since it is not possible to maintain a differential signal path to headphones[4]. The performance to the headphones is nevertheless very high as shown here in this standard $1kHz$ 16 bit FFT

[4]There is no industry standard means to drive stereo Headphones differentially. The three terminal TRS connector standard allows only Left, Right and Common wires to the head set.

3.9 Full Spectrum Jitter Test

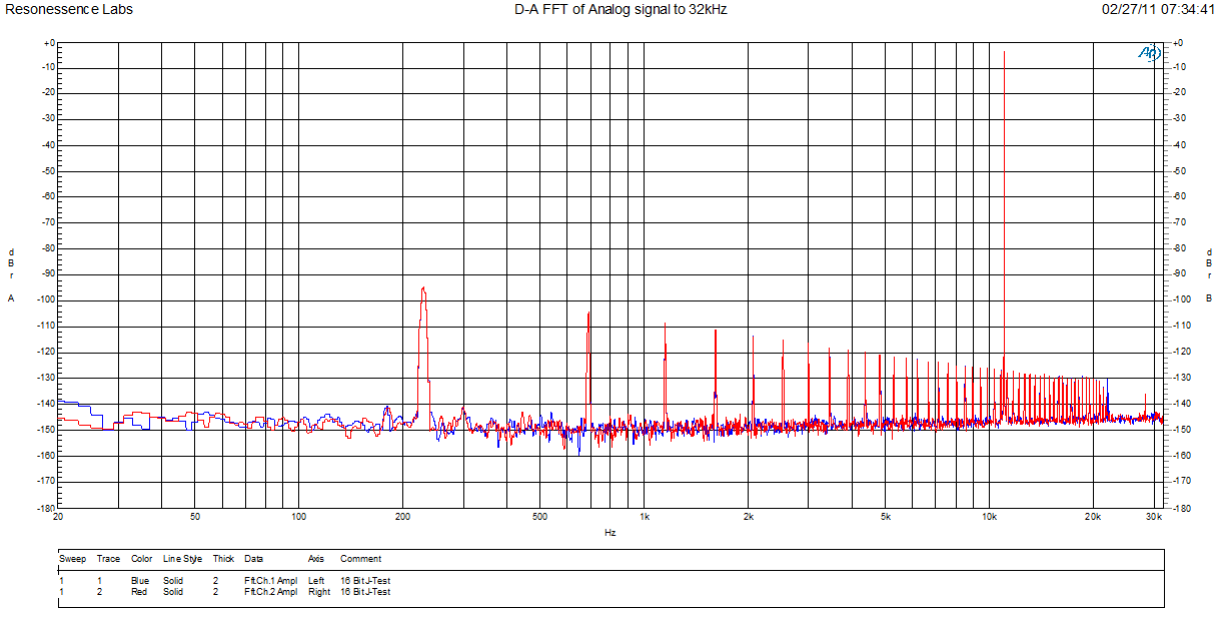

Figure 3.10: Jitter Test - full spectrum

Jitter is the error in a timing signal: jitter in a DAC means that the digital data is reproduced at the wrong time. This can be heard as a "muddiness" or lack of precision in the sound stage.

After THD and Noise, the parameter that has a large effect on perceived quality is the jitter. Jitter refers to the variation in the time of the digital sample.

To test for jitter, a carefully constructed series of sample points are passed to the DAC. These sample points are at exactly the sample rate (often notated as fs) divided by four, and imposed upon them is a variation in just the least significant bit (LSB) at about $230Hz$[5]. Consequently the FFT has a large signal at $11.025kHz$ and a tiny square wave signal[6].

Jitter may be assessed by inspecting the widening (i.e. "spreading" at the base) of the main tone and any widening of the harmonics of the $230Hz$ signal. The above graph confirms the presence of the $230Hz$ square wave at $-90dB$, but does not show sufficient detail of the $11.025kHz$ signal. On the next page more detail is provided.

[5]The low level tone is actually at $fs/192$ or 229.6875Hz

[6]The square wave signal measures just below $-90dB$ because the LSB is taken to be the LSB at the 16 bit data width – again as from a CD. The "real" LSB of the Invicta system is that of a 24bit data width, but this Jitter test is conventionally done at the 16 bit level. Refer to [1] for details.

3.10 Reduced Spectrum Jitter Test at 16bits

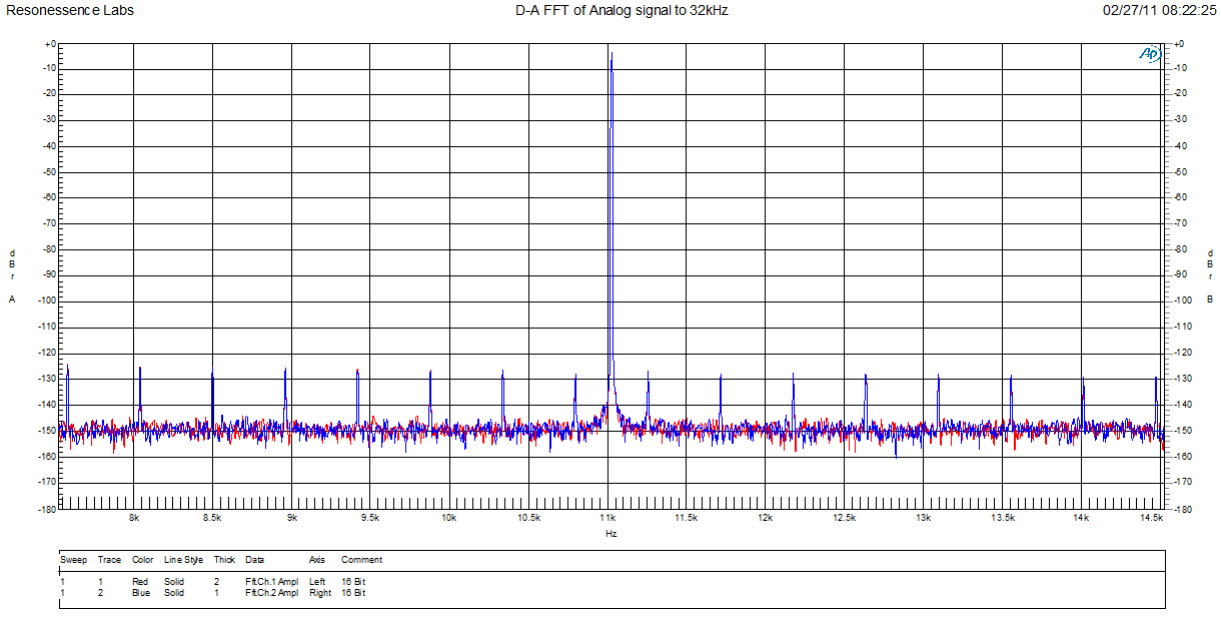
Resonessence Labs
D-A FFT of Analog signal to 32kHz
02/27/11 08:22:25
dBr A
dBr B
+0 -10 -20 -30 -40 -50 -60 -70 -80 -90 -100 -110 -120 -130 -140 -150 -160 -170 -180
8k 8.5k 9k 9.5k 10k 10.5k 11k 11.5k 12k 12.5k 13k 13.5k 14k 14.5k
Hz
Sweep Trace Color Line Style Thick Data Axis Comment
1 1 Red Solid 2 FftCh.1 Ampl Left 16 Bit
1 2 Blue Solid 1 FftCh.2 Ampl Right 16 Bit

Figure 3.11: Jitter Test - detail in $fs/4$ region 16 bit data

To see the effect of jitter, this graph is the same as the one on the previous page but zoomed-in to the region of $fs/4$ or about $11kHz$.

Jitter on the internal clock of the DAC causes a widening or spreading at the base of the main tone. That spreading may be significant in some DACs - the bottom of the $11kHz$ tone may spread to cover the two harmonics of the $230Hz$ tone that are nearby.

This data shows very little spreading at the base of the main tone – nowhere near impinging upon the adjacent harmonics of the $230Hz$ signal. This shows that the jitter is exceptionally low.

The main tone is very well resolved: there is very little indication of "spreading" which would indicate jitter.

3.11 Reduced Spectrum Jitter Test at 24bits

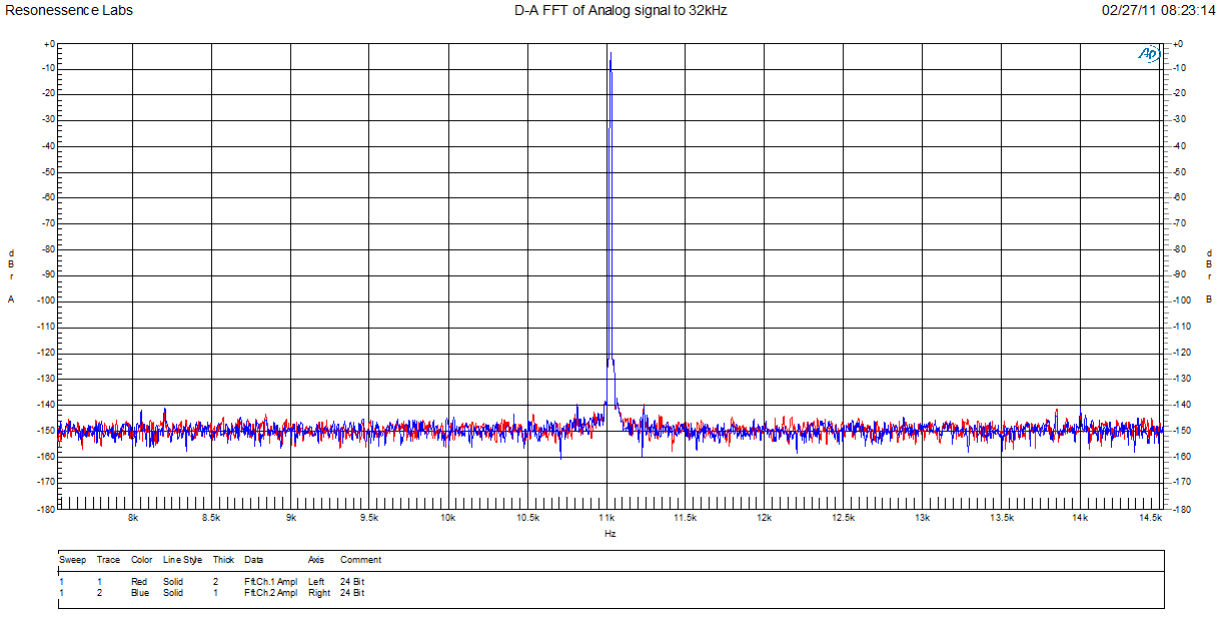

Figure 3.12: Jitter Test - detail in $fs/4$ region 24 bit data

The data width does not affect the jitter test: both 16 and 24 bits show excellent jitter specification.

This is the jitter test for 24 bit data. It shows the same result as at 16 bits; there is very little indication of spreading in the tone. Note the interesting thing about the jitter test set up: because the data in the digital stream is phase locked to the sample rate(that is to say, the encoded frequency is an exact division of the sample rate, $fs/4$ in this case) there is no quantization noise present, to any arbitrary level in the digital data[7]. The fact that this is 24 bits has made no difference to the result, as expected.

[7]This absence of quantization noise is not particularly surprising: any square wave encoded in the data stream will exercise only two codes (the upper and the lower level of that square wave). Furthermore, if the frequency is an exact division of the sample rate, the same pattern of bits will emerge from the band-limiting filter over and over again. Therefore, there is no mechanism to spread the quantization energy into a wide band and the noise floor is that inherent to the DAC itself, independent of the bit width.

3.12 Jitter Test Detail

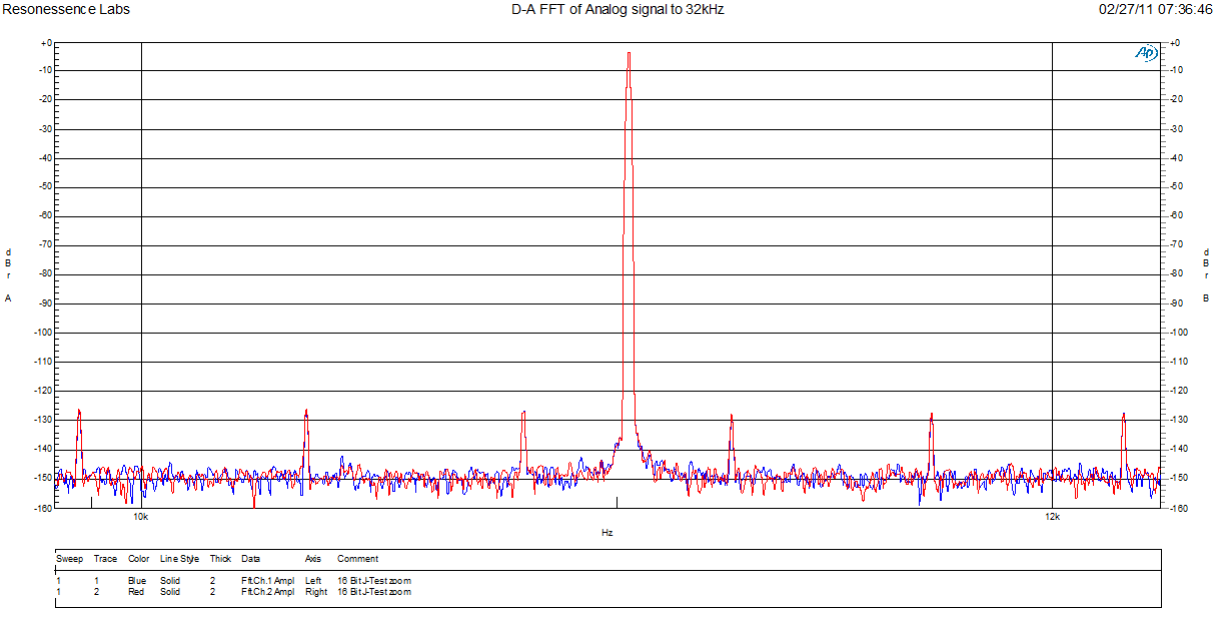
Resonessence Labs
D-A FFT of Analog signal to 32kHz
02/27/11 07:36:46
dBr A
dBr B
10k
12k
Hz
Sweep Trace Color Line Style Thick Data Axis Comment
1 1 Blue Solid 2 FftCh.1 Ampl Left 16 BitJ-Test zoom
1 2 Red Solid 2 FftCh.2 Ampl Right 16 BitJ-Test zoom

Figure 3.13: Jitter Test - detail in $fs/4$ region of the spectrum

This is the Jitter test in the region of the $11.025kHz$ signal. Note the width of the main signal: it is narrow and nowhere near intruding onto the surrounding $220Hz$ harmonics.This indicates that the Invicta internal reconstruction clock is good. It exhibits very low phase noise.

We can assume that the recording studio has an essentially perfect timing source (often referred to as a "clock" in the electronic system) and the digital samples have been taken precisely where they should have been in time.

If, however, the DAC used to recreate the analog signal has a badly defined clock – a badly defined timing source – then the precision is lost. Remarkably perhaps, listening tests show that the skilled audiophile will be able to hear a degradation in the sound quality unless the DAC clock is exceptionally good.

The DAC clock is an example of where engineering judgment and audiophile experience diverge. The engineer is inclined to think that jitter of 0.1% of the unit interval (the sample rate) is a good clock. Listening tests prove that the audiophile can hear artifacts down to $< 0.001\%$ of the unit interval.

The narrow width of the main signal shows that the Invicta has an excellent reconstruction clock and will not suffer from phase noise induced quality reduction.

(Refer to [1] for many more notes on jitter).

3.13 Frequency Response at 44.1kS/s

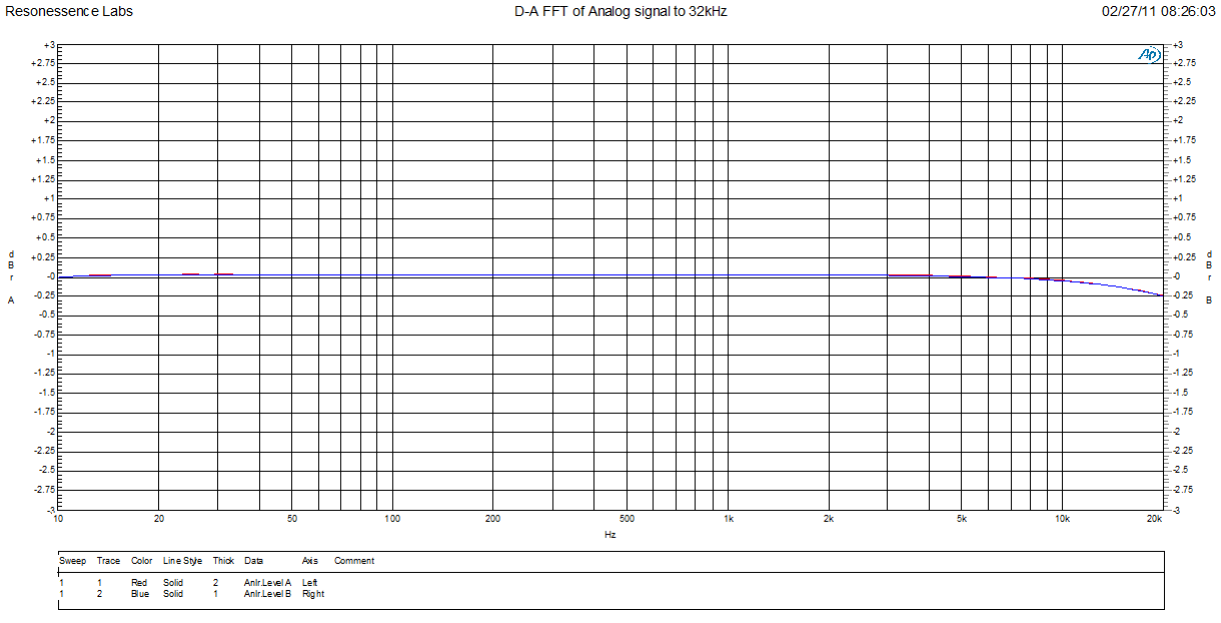

Figure 3.14: Frequency Response at $44.1kS/s$

The Invicta has less than $-1dB$ roll-off with the $20kHz$ band for $44.1kS/s$ rate.
In fact, the roll-off is measured to be $-0.25dB$.

In the $44.1kHz$ sample rate, $20kHz$ is the target maximum audio bandwidth[8]. Within that bandwidth, the system ideally has a perfectly flat frequency response.

This graph shows the Invicta system has about $0.25db$ roll off at $20kHz$ – well within the acceptable limit.

[8]From a theoretical point of view, the digital data has an absolute limit of $22.05kHz$, but this mathematical limit can never be reached because the band-limiting filter to remove the adjacent alias signal would have to be infinitely steep.

3.14 Frequency Response at 192kS/s

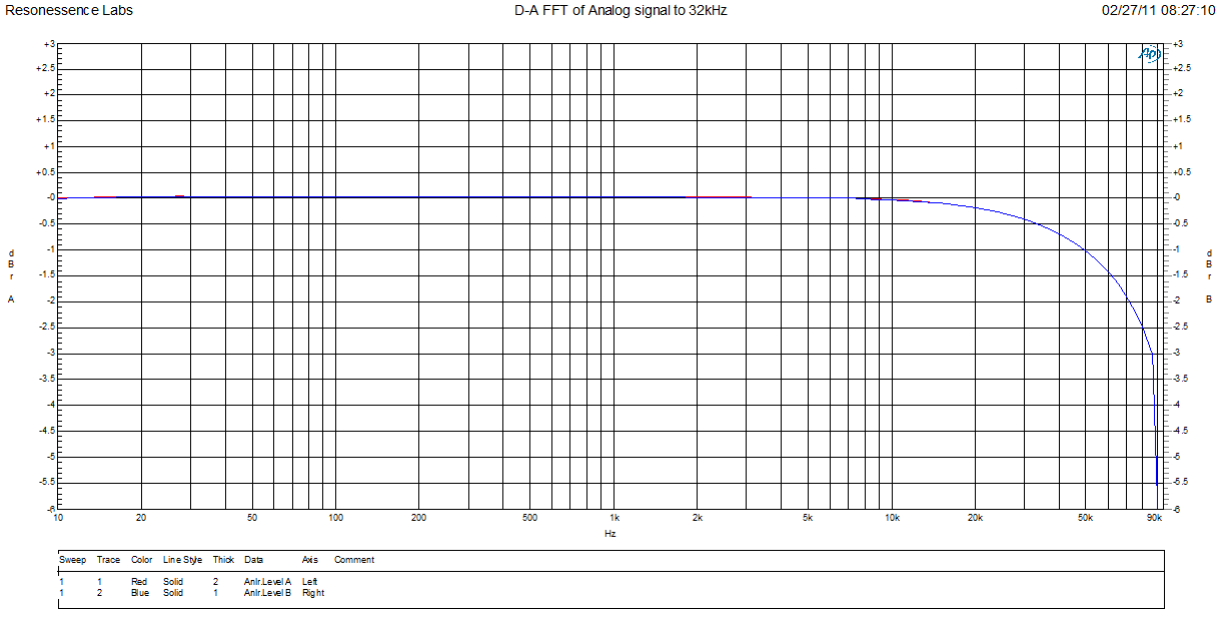

Figure 3.15: Frequency Response at 192.kS/s

As mentioned in the reference [3], it is now known that the human ear delivers sounds of up to at least $50kHz$ to the brain. While not consciously reported as perceptible, such sounds may nevertheless contribute to the listening experience – there is some evidence from listening tests to support this assertion.

When provided with $192kS/s$ data rate the Invicta achieves a $-1dB$ bandwidth of $50kHz$ and a $-3dB$ bandwidth of over $80kHz$.

Fortunately, advancing technology has brought higher sample rates and higher resolution to the audiophile: music can now be acquired in 24bit format with as high as $192kS/s$ data rate. The common mathematical limit of $22.05kHz$ is now broken since these data rates can support signals to $80kHz$ or higher.

This graph shows that the Invicta has a $-3dB$ bandwidth exceeding $80kHz$.

3.15 DNR vs DC Offset

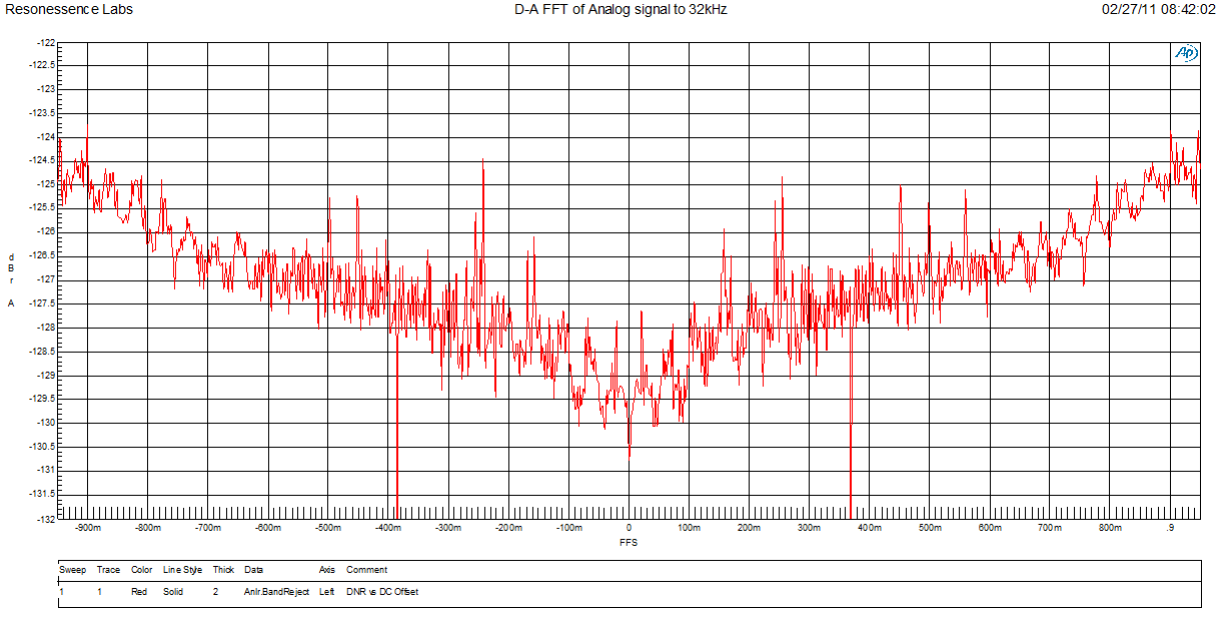

Figure 3.16: Dynamic Range vs. DC Input

A phenomenon seen in Noise Shaping Modulators is tightly controlled in the Invicta: it has very little degradation of DNR with DC input.

Noise shaping control loops ($\Sigma\Delta$ modulators and the HyperStream modulator in the ESS Sabre DAC used in Invicta), exhibit an interesting phenomenon: the dynamic range measured as the signal to noise ratio for a quiet input is not independent of the DC input condition[9].

The change in DNR with DC Offset can be quite profound; as much as a degradation of $20dB$. This graph shows that the Invicta and its internal ESS DAC exhibit a small degradation of about $-6dB$ for even a large (more than 90%) DC offset.

[9] In fact, as reference [2] explains, the phenomenon extends to more than just the DC offset, but to any constant derivative applied as input to the noise shaping loop.

3.16 Linearity in 24bit Mode

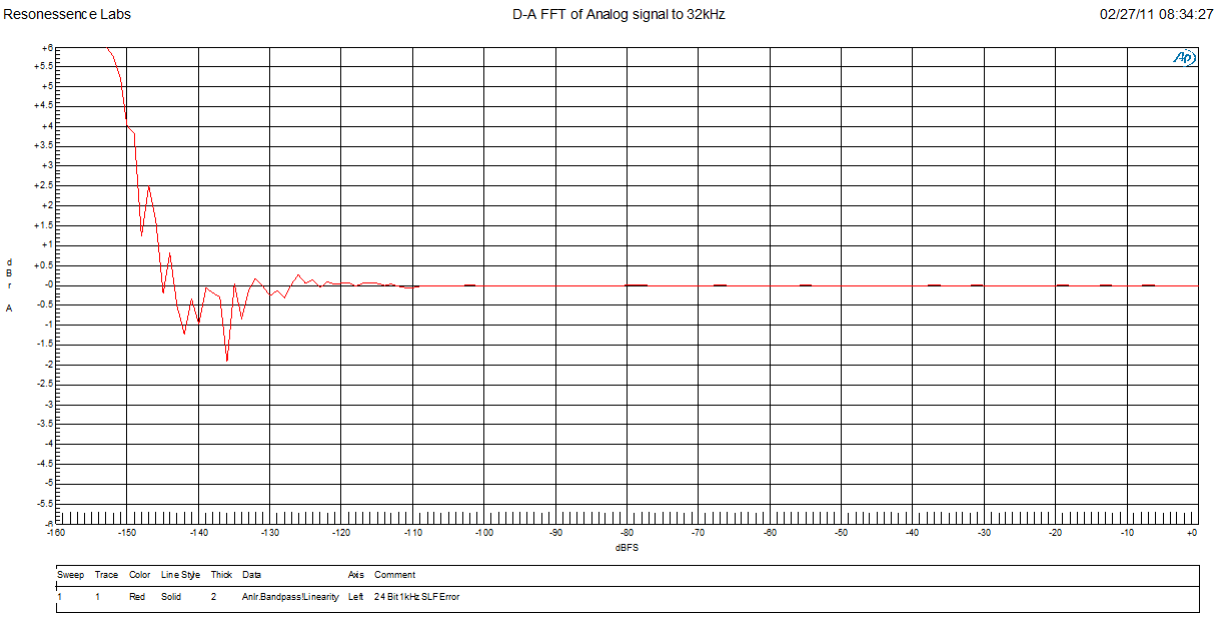
Resonessence Labs
D-A FFT of Analog signal to 32kHz
02/27/11 08:34:27
dBr A
dBFS
Sweep Trace Color Line Style Thick Data Axis Comment
1 1 Red Solid 2 Anlr.Bandpass\Linearity Left 24 Bit 1kHz SLF Error

Figure 3.17: Linearity of Response with Changing Input Amplitude

Linearity, when applied to a DAC, potentially has two meanings. The first, used in instrumentation, interprets "linearity" as the degree to which the DC transfer characteristic (from linear input code to output parameter) corresponds to a straight line. However, "linearity" to the audio engineer describes how the output amplitude at some given (non-zero) frequency matches that of the applied input amplitude. The two should match exactly, and the deviation of measured output amplitude from the ideal straight line is shown in this graph.

The output amplitude matches the input amplitude very accurately; down to $-130dB$. This is a very good result.

The output amplitude matches that expected from the input amplitude within a tiny fraction of a dB from $0dB$ all the way down to $-130dB$.

3.17 Time Domain Response -90dB

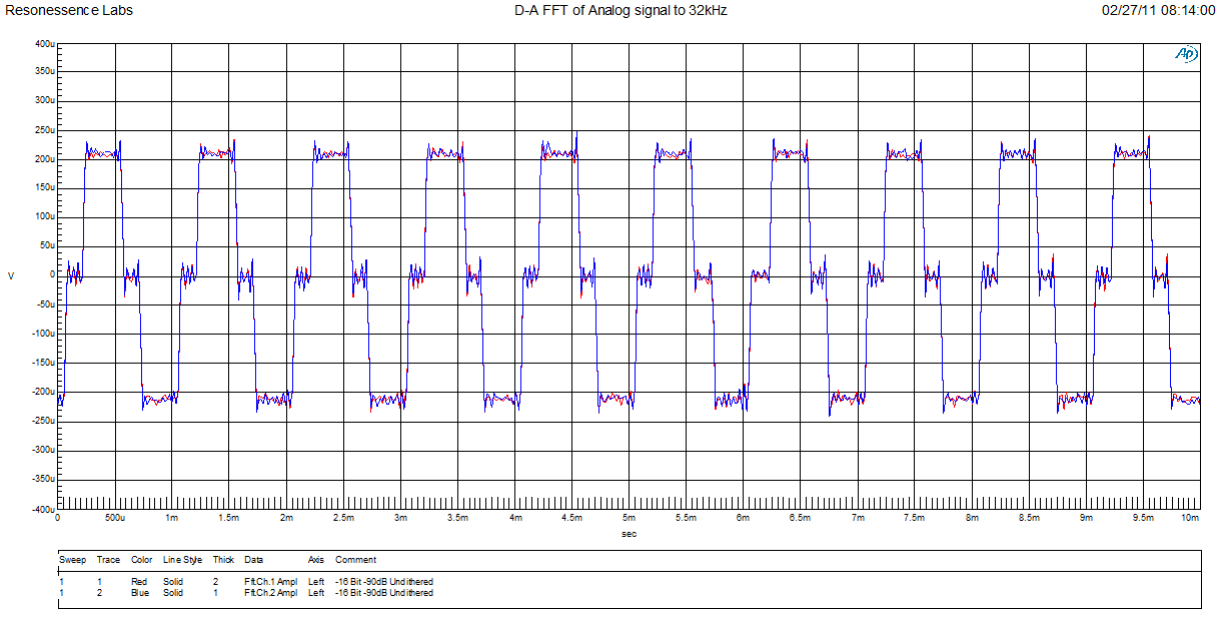
Resonessence Labs
D-A FFT of Analog signal to 32kHz
02/27/11 08:14:00
V
sec
Sweep Trace Color Line Style Thick Data Axis Comment
1 1 Red Solid 2 FftCh.1 Ampl Left -16 Bit -90dB Undithered
1 2 Blue Solid 1 FftCh.2 Ampl Left -16 Bit -90dB Undithered

Figure 3.18: Time Domain Response for Small Signal

The time domain response resolves a small signal (1LSB at 16 bits) and has a symmetrical response.

This graph shows the Invicta response to a small, three-level signal applied to the input. The signal is $\pm 1 LSB$ around zero at 16 bits.

Despite the small signal, the output is clearly resolved and the waveform noise is low and the response symmetrical.

3.18 Inter-modulation Distortion Test

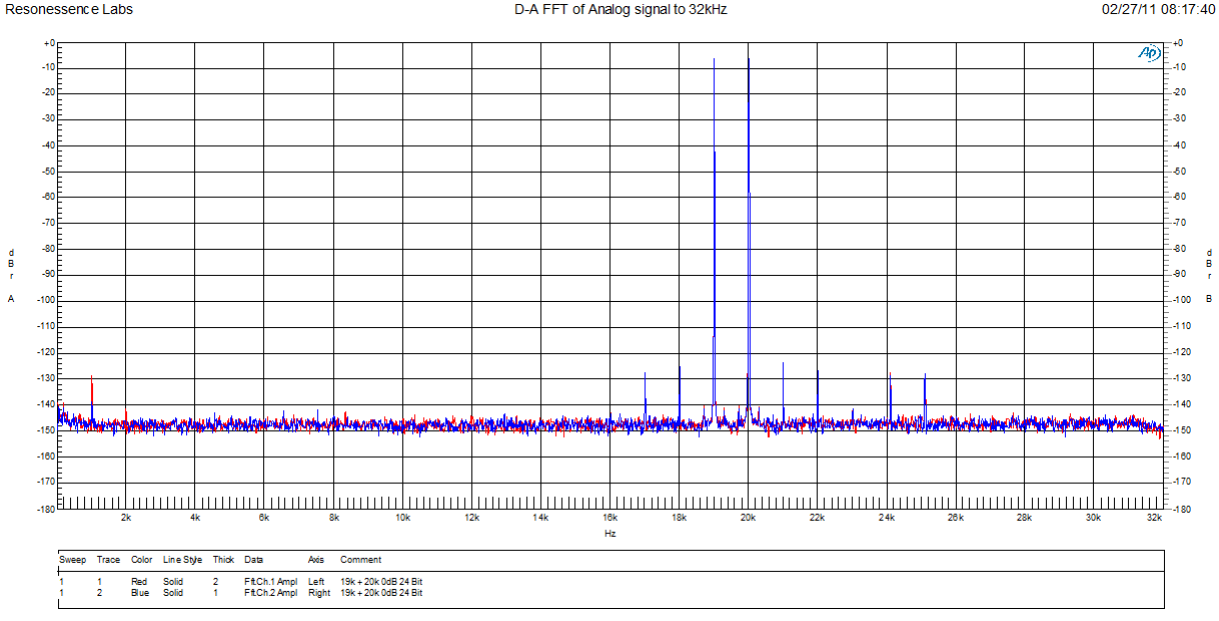
Resonessence Labs
D-A FFT of Analog signal to 32kHz
02/27/11 08:17:40
dBr A
dBr B
Hz
Sweep Trace Color Line Style Thick Data Axis Comment
1 1 Red Solid 2 FftCh.1 Ampl Left 19k + 20k 0dB 24 Bit
1 2 Blue Solid 1 FftCh.2 Ampl Right 19k + 20k 0dB 24 Bit

Figure 3.19: Inter-modulation Distortion

Harmonic Distortion of a single input tone gives rise to spurious signals at multiples of the input frequency. For example, a $1kHz$ signal will generate a second harmonic at $2kHz$, a third harmonic at $3kHz$ etc. In an audio DAC, the bandwidth is limited, typically to $20kHz$, and so a simple test for harmonic distortion above $10kHz$ will not generate visible errors: all the harmonics are $20kHz$ or more, hence outside the bandwidth and not seen.

Inter-modulation products are small in the Invicta, being less than $-120dB$. They are no more than the distortion products measured at low frequencies.

Although invisible to a single tone, harmonic distortion will "mix" (that is, mathematically multiply) with any other signal in the input and can be made to create in-band products that are visible. Specifically, the third order products (such as $2f1 - f2$ and $2f2 - f1$) cause distortion products adjacent to the frequencies $f1$ and $f2$, as are visible in the graph above.

In the graph above, $19kHz$ and $20kHz$ tones are applied simultaneously and distortion products at $17kHz$, $18kHz$, $21kHz$ and $22kHz$ are visible. These spurious tones show the distortion of the $19kHz$ and $20kHz$ signals even though those distortion products are far out of band.

3.19 Swept Frequency Low Level Noise Test

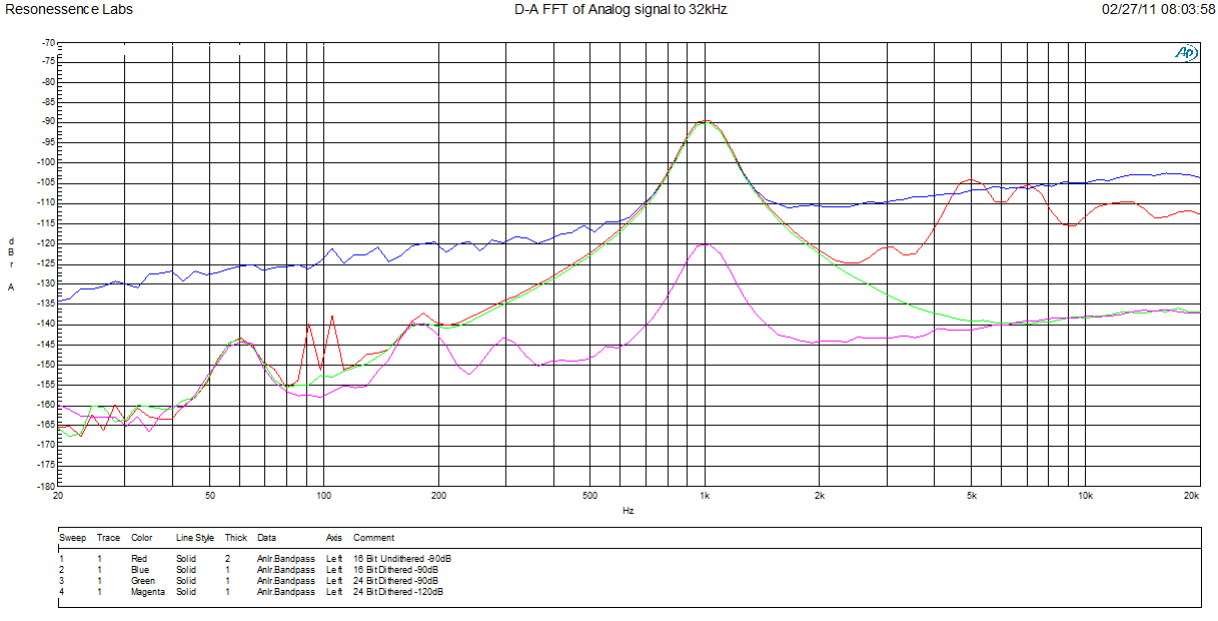

Figure 3.20: Frequency Sweep Noise Test

The noise level of Invicta can be explored as a filter passes over a very low level tone. Once the filter moves off the tone, the noise drops quickly to a low level indicating a very low noise.

The noise of the DAC may be assessed by injecting a certain low level signal and sweeping a narrow band filter through that signal. In the ideal case, if the filter were perfect and the DAC had no noise, there would be no output signal at all until the filter passed over the low level signal frequency.

The results on this graph approximate that situation. In three cases the input signal was $-90dB$ and in one case the input signal was $-120dB$. In all cases the peak signal is the expected one, either $-90dB$ or $-120dB$ as input[10].

[10] Refer to various articles in Stereophile magazine for a more detailed explanation of this somewhat specialized test.

3.20 THD+N vs Frequency (in dB)

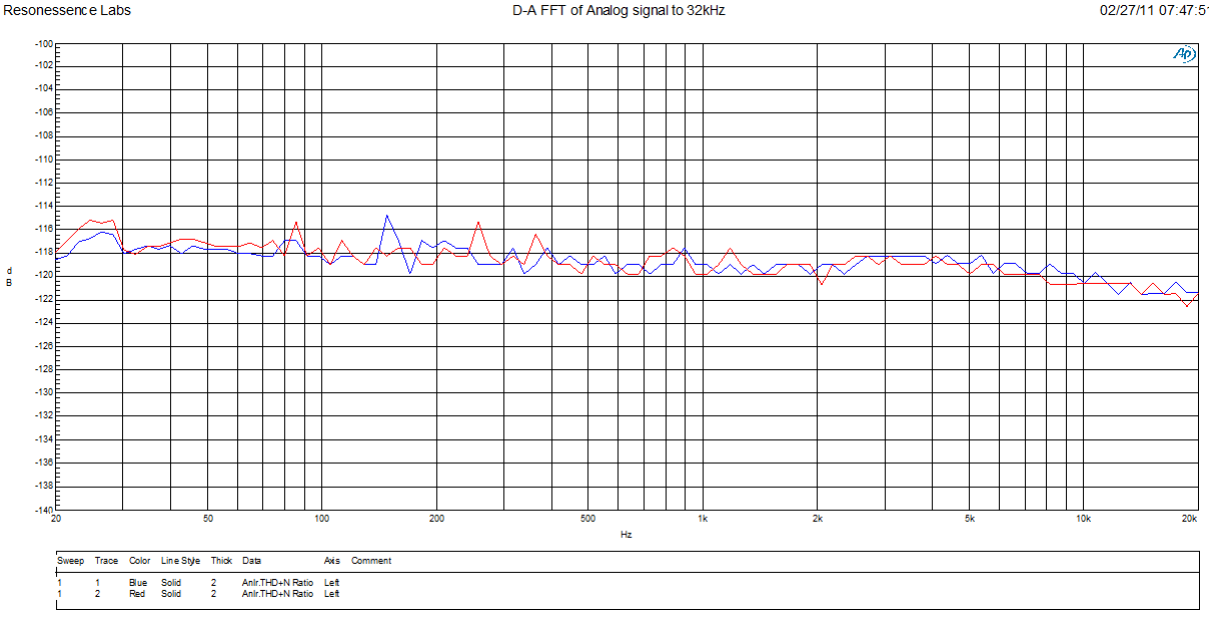

Figure 3.21: THD+N vs. Frequency (in dB)

Total Harmonic Distortion plus Noise can be conveniently measured if a sharp notch (band reject) filter is placed over the input tone. The filter will remove the input tone and the RMS value of everything out of the filter then represents both the noise and any harmonics present[11].

The Invicta THD+N never exceeds $-114dB$ for all frequencies in the audio band.

If an input tone is swept across the audio band and a similarly swept notch filter tracks that signal, then the output will be the THD+N vs Frequency. This is the method used here and the resulting graph shows how distortion and noise vary with frequency[12].

[11] Because of this convenient method – namely the notch filter over the input tone – THD+N is much easier to measure than THD on its own. This is why most audio measurements report THD+N and not just THD.

[12] This is an important test because certain forms of DAC may show an anomalously low THD+N in the $1kHz$ region where THD+N is conventionally measured.

3.21 THD+N vs Frequency (in percent)

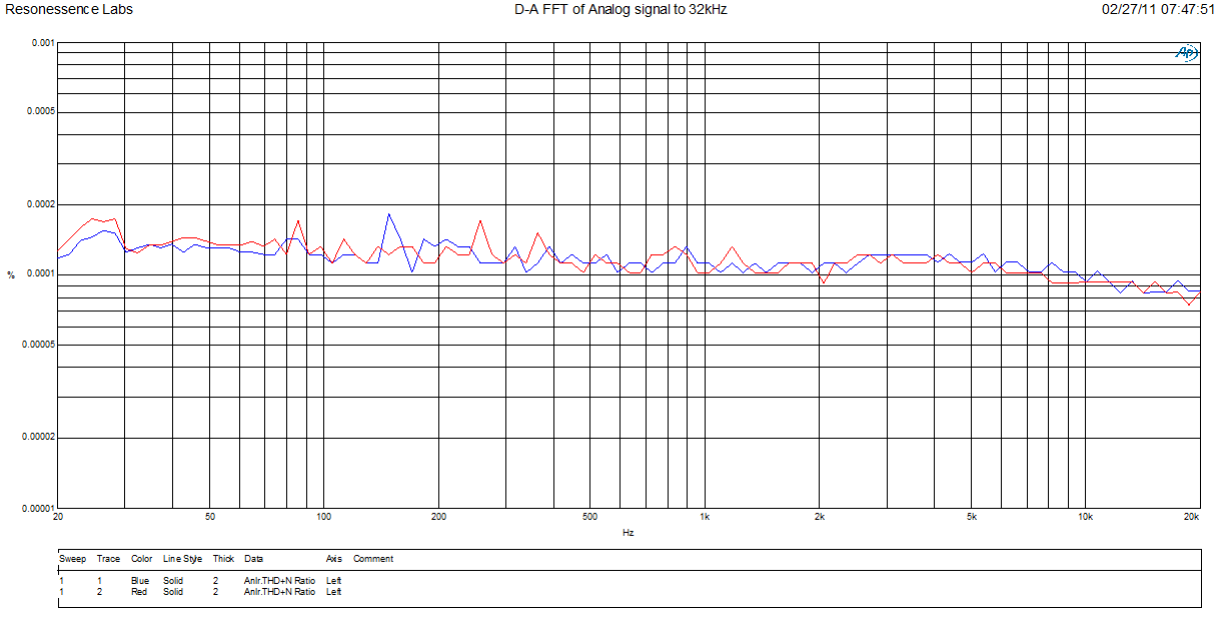

Figure 3.22: THD+N vs. Frequency (in percent)

The THD+N of the Invicta is less than 0.0002% for all frequencies in the audio band.

THD+N is also commonly reported as a percentage figure. This was appropriate when the state-of-the-art created THD+N numbers just below the 1% figure, but today, since technology has improved, THD+N in percent is less helpful: it has become a matter of carefully counting the zeros before the significant fraction.

However, to be complete, this is the THD+N vs. frequency as measured in percentage of the applied signal.

3.22 Cross-Talk vs Frequency

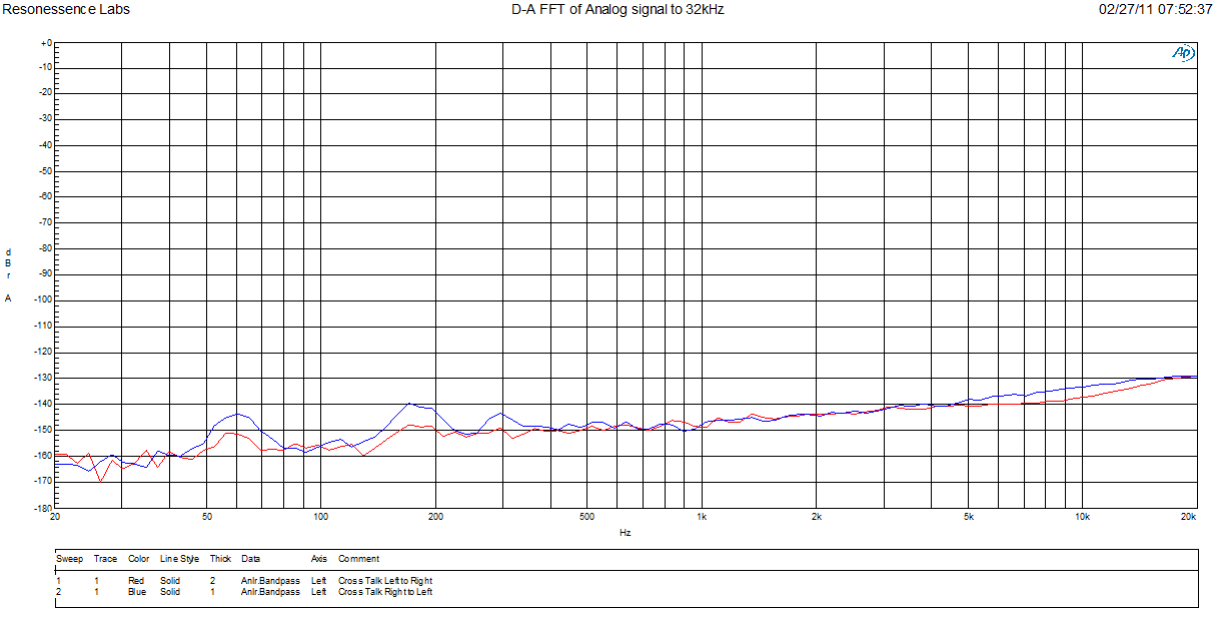

Figure 3.23: Cross-talk vs Frequency

Cross-talk is the undesirable tendency of one data channel (for example, the Left channel) to "feed-through" or cross-talk into the other channel (Right channel). This is an important parameter for those audio systems that recover the stereo information from certain compound signals (such as the V-groove in a vinyl recording, or the multiplex signal in an FM transmission), but in modern digital technology the data source can be assumed to have a perfect separation of channels.

Cross talk in the Invicta is excellent: it remains below $-130dB$ for all frequencies.

Measuring cross-talk in a modern digital system such as a DAC will expose any failure in the design to retain this separation of digital data. Cross talk in a modern system reflects a finite power supply rejection, or a less than perfect arrangement of ground signals in the system.

The graph above shows cross-talk vs. frequency. Its slight rise suggests a capacitive coupling; nevertheless, this measurement of no worse than $-130dB$ even at $20kHz$ is a good result.

Chapter 4

Notes on Audio Engineering

4.1 The Sense of Hearing

We have evolved the remarkable sense of hearing over millions of years. Each of us has the ability to detect variations in pressure traveling as waves in the air around us. The human ear is an astonishing sensor of these sound pressure waves: the pressure difference may cause our eardrum to move less than the diameter of a hydrogen atom, and yet this tiny disturbance is accurately delivered to our conscious mind to allow us to act upon it. The dynamic range of hearing is breathtaking, being very much higher than the range of vision. The ratio of the smallest perceptible energy to the highest tolerable energy is about 130dB, which is ten trillion to one or 10^{13}.

Experiments show that the ear delivers information to the brain up to 50kHz, not 20kHz as commonly assumed.

It is commonly thought that the human ear detects sounds in the frequency range of a few Hertz to 20kHz. This is supported by experiments where test subjects are asked, "Can you hear this sound?" and 20kHz is the highest frequency where a person with good hearing will report that they can hear it. However, we now know that this is misleading: if a subject is placed in a brain scanning device we can show that the brain is responding to sounds up to about 50kHz.[3] It is simply that the conscious mind is not aware of sounds above 20kHz but something in the brain does register the higher frequencies. Although not presented to the conscious mind as an audible tone, the presence or absence of higher frequencies does have a measurable effect, and we may speculate

that this phenomenon is one source of the listener's perception that a system resolving these higher frequencies "sounds better" in listening tests than one that renders only 20kHz of bandwidth.

The ear is an active device: each hair of the Cochlea that responds to sound, also has muscle cell attached that the brain uses to "tune" and adapt the hearing process.

Another common opinion now proved wrong, is that the ear is a passive device similar to a microphone, faithfully translating sound pressure into some internal format that the brain can use. In fact the ear's interaction with the brain is far more complex. The cochlear is an active amplifying device[4, 5] consisting of at least two mechanisms: the so-called outer hair cells and the inner hair cells. The outer hair cells are known to be adaptive, each has a little "motor"[6] attached that tunes the reception and this is how the ear achieves its remarkable frequency resolution - a resolution far higher than a set of simple resonators could achieve[7]. We may speculate that the active nature of the hearing process supports the conclusion that audiophiles have an ability to sensitize their hearing, the brain taking an active role in the listening process.

The sophistication of human hearing far exceeds that of other animals, bats being a notable exception. Some animals hear higher frequencies. Birds and frogs are known to posses specific neural processing of audio signals to find mating calls buried in what would otherwise be random noise. But the human sense of hearing is remarkable: the ear and brain are one, perception arises in the higher evolved regions of the brain where audio signals have been "spike encoded"[8] and filtered into elemental representations of music and spoken words. It is likely that the human ability to speak arises at this higher level where the mind is working on significantly pre-processed signals. We may speculate that music appreciation also operates at this higher level. An imperfection or fault in rendition of recoded music will propagate up to this higher level and be perceived as "something wrong" with the sound, but the detail of that error has been suppressed by the unconscious pre-processing. This will give rise to apparently meaningless assertions of the audiophile such as, "I listen to the 'gaps' between the sounds" or, "it is the noise 'behind' the transient that is important".

Processing of sounds takes place below the level of conscious awareness.

In these higher centers of the brain the sounds and sights are integrated into one whole, and surprisingly perhaps, experiments show that the eyes can effect the ears and cause the ears to "line up" the three dimensional audio image they create with the eyes[9]. This is no doubt why audiophiles commonly close their eyes to appreciate fine music.

The ear's ability to resolve signals in time is equally remarkable: we can easily calculate that a resolution angle of five degrees in sound location corresponds to less the $50\mu S$ in time difference[1]. Clearly, therefore, the brain is able to process at least the difference in arrival time of audio signals on a time scale significantly less than $100\mu S$.

Given this degree of sophistication in our sense of hearing, we can only marvel at the fact that audio engineers can record sound at all. Who would imagine that the vibrations of a speaker cone, driven through necessarily imperfect electronics, picking up a signal from a transducer following a v-shaped scratch in the surface of a vinyl record would work as well as it does? Who would imagine that a microphone could do a tolerable job of transposing sound pressure into an electronic signal that meets the quality level the ear expects? Perhaps, in fact, none of these things (vinyl records, speakers, microphones) work well at all - perhaps all the magic lies in the ear which extrapolates across errors, fills in imperfections and delivers to our conscious mind a remarkably consistent audio picture where a great many "details" are actually psycho-acoustic "patches" over an imperfect signal.

The hearing process is able to extrapolate missing details.

There is a great deal to support this latter hypothesis. The existence of MP3 players essentially proves it: in MP3 encoding the sound signal is perversely distorted from its pure original, but the perversion is carefully constructed to match the ear's ability to extrapolate detail, to fill in the missing structure. There can be no surprises, no unexpected detail in MP3 music - they are simply not there. You will not be moved to comment on fine detail in the MP3 playback, that detail is in your head - your own ultra-sophisticated audio processing has cleaned up the MP3 signal to an acceptable level.

However, there is an even more remarkable example of the pre-processing power of the human ear. In our modern world we have moved on from the outstanding technology of the Victorian era that brought us color TV via electrons moving in the vacuum of a CRT tube, to the visually more stunning (but technically lower resolution) image on Plasma or LCD TVs. These new TVs appeal due to their flatness: we hang them conveniently as we might hang a picture on our wall.

However, flat TVs do not provide enough room to include a quality

[1] Assuming the speed of sound is 340m/s and the ears are separated by 15cm, then for a 5°angle resolution we have an additional path length of $0.013 = 0.15 * sin(2\pi\dot{5}/360)$ meters which represents 0.013/340 seconds or $38\mu S$.

loud-speaker. The low end (sub 200Hz) response of a typical flat TV loud-speaker is not just poor, it is almost completely absent. It is mechanically unable to re-create a sound less than 200Hz or so without significant attenuation. To try to compensate for this, some early designs boosted the bass response causing the drive signal to rise as the frequency rolled off. But this does not work - the speaker simply starts to distort more and it is not a viable solution.

Certain forms of "Bass Enhancement" create acoustic illusions that fool the brain into thinking tones are present when they are not.

The solution that is now used in next-generation small loudspeaker design exploits the sophisticated pre-processing of the human ear. It is found that when a low frequency signal of say 100Hz is presented to a listener he will, of course, respond that he hears 100Hz. But astonishingly, if the fundamental of that 100Hz is removed, leaving only its harmonics, the listener will still report that he hears 100Hz, despite the fact that no signal at 100Hz is present! This is a testament to the power of the ear and brain - something is profoundly wrong with the signal received by the ear: it is missing its fundamental. But the brain and ear "know" that it must be there because its harmonics are present - and remarkably, below the level of the conscious mind, this missing fundamental is added back into the signal and the conscious mind hears the 100Hz signal[2]. The TV DSP engine implements this psycho-acoustic "trick" and creates appropriate harmonics to send to the speaker. The speaker is not required to process any signal less than about 200Hz, but the listener hears the sound of 100Hz or lower emerging from the speaker. It is a remarkable example in the audio domain of a phenomenon used extensively in the video world: the electronics is exploiting what is known of the brain's processing of the senses and this creates effective and commercially desirable home electronics.

The Audiophile, the lover of fine music, is able to attune his or her hearing to a degree above the inexperienced listener, and can indeed hear differences between various DAC technologies.

All this understanding of the process of hearing should help us appreciate the audiophile: when he tells us that he can hear the difference between a sigma-delta modulator and a non-oversample DAC, we should not be dismissive. It is likely that he is attuned to some subtle aspect

[2]The knowledgeable reader will question this: how can the signal pattern from the 100Hz with missing fundamental be distinguished from a signal of 300Hz (the first harmonic actually present)? The answer is that it can because the relative amplitude of the harmonics is different. The 100Hz with no fundamental has a tone at 500Hz (its fifth harmonic) in addition to its 300Hz tone of the third harmonic. But the real signal at 300Hz has no 500Hz component: its third harmonic (the first one) is at 900Hz. The brain can apparently easily recognize and sub-consciously sort out this difference. The 100Hz with no fundamental is perceived as 100Hz, the 300Hz is perceived as 300Hz, despite the fact they share a common 300Hz tone.

of the hearing process that we may miss, and any amount of comparing typical audio measurements (frequency response, THD, etc) may not capture the distinction.

In fact, the typical technical measurements do not capture the subtleties. Many years of testing, many trials of various techniques have taught us, here at Resonessence Laboratories, that second and third order effects (over an above the common measurements) are what make a great audio DAC as opposed to a good audio DAC.

Both is better: technical excellence and that attention to the audiophile that make an outstanding DAC.

First we excel at the common specifications, then we endeavor to excel in the areas that make a great DAC. We do not make devices that exploit any psycho-acoustic artifacts, rather we aim to deliver to the audiophile ear the most perfect rendition of the sound that is technically possible. The audiophile does not listen to us - we listen and learn from the audiophile.

4.2 About Common DAC Specifications

Audio engineers strive to please the customer and deliver verified high quality components to the discerning listener. To differentiate their products (and perhaps justify a higher selling price), the engineer will document the performance of the product using industry standard instrumentation. Consequently, specifications provided are those that the instrumentation can measure: if the machine cannot measure it, it cannot be listed in the technical specifications table. This is obviously scientifically justified, since otherwise the competition to create the best audio devices would degenerate into matters of opinion rather than of verifiable measurements.

Read the Specifications, but then listen to the Invicta through your favorite amplifier or headphones.

But what is the best course of action when the audio engineer is persuaded that a scientifically verifiable difference (in double-blind testing) does actually exist and that this does not correlate to any technical measurements? This is a difficult situation since this difference between the good and the excellent audio product cannot be reduced to a specification. Are we simply to trust that the respected customer (or reviewer) has no bias and does posses the hearing acuity needed to appreciate our products? The answer, in the current state of the measurement art, is yes - that is exactly what we are going to do - the final judgment lies in the ear of the audiophile.

However, Resonessence would like to promote development of the audio measurement art and to this end share our current understanding of where technical measurements may fail to capture the audiophile's perception of quality. Please be assured that the following notes on why these technical measurements may fail to do justice to the listener is in no way designed to justify a less than perfect technical measurement of the Invicta DAC: you will find the Invicta to excel in the technical measurements.

4.2.1 Decibels (dB) in Audio Measurements

The decibel is a logarithmic unit and is one tenth of a bel, which indicates a factor of 10 in power. $dB = \frac{log_{10}(P1/P2))}{10}$.

The range of sounds that the human ear can perceive is staggering. The loudest sound (which is painful) has more than 1,000,000,000,000 times more energy that the quietest sound (which is just perceptible in ideal conditions). For such a large dynamic range a logarithmic scale is appropriate and audio engineers use a term developed in the early 20th century called the "decibel", which is one tenth of a "bel". A "bel" is never used on its own, even from its earliest conception, the decibel (a tenth of one bel) was proposed as the basic unit since this was approximately the loss in signal over one mile of telephone cable and was thought to be the minimum change in amplitude that a listener would notice. One bel is a ratio of ten to one, and so one decibel is a ratio of $10^{0.1}$ in power.

Given that one decibel (dB) is a tenth of the logarithm of the power, when we review a graph showing performance of an audio system we can immediately conclude that power changes by a factor of ten for every ten decibels. For example, should we decide to listen to our favorite rock band at $0dB$ and then to our favorite classical guitarist at $-40dB$, the difference in energy delivered to our ear is 10,000 to 1, because $-40dB$ is a ratio of $10^{40/10}$ or 10^4 or 10,000 to 1. And $-40dB$ is not that quiet: unless we have ruined our ears in early life actually listening to rock at $0dB$[3], we will be able to hear to maybe $-90db$, which is an energy ratio to the $0dB$ rock band of 1,000,000,000 to 1.

In the electronic system, the voltage (which is usually the parameter processed as analogous to the displacement of the microphone and speaker diaphragm in the electronic representation of music) encodes not the energy, but the square root of energy. Consequently, when the

[3]Which the author, sadly, has done.

audio engineer considers decibels he has in mind that $20dB$ (not $10dB$) corresponds to a factor of ten in voltage. This means that when the audio DAC is processing, for example, $0dB$ as 1V rms, it will process $-40dB$ as one hundredth of 1V rms or 10mV rms. We may rightly conclude that in order to achieve the $-130dB$ DNR of the Invicta, the 4V rms $0dB$ output must have a voltage noise of less than 1.25μV rms since this is $4 * 10^{-130/20}$.

dB's can be confusing: they actually refer to power and ten of them make a factor of ten in power level. But electronic audio devices process volts and energy is volts squared, so in volts it takes $20dB$ to make every factor of ten!

In audio measurements of a DAC such as the Invicta, dB, a dimensionless unit is expressing the ratio to the full scale that the DAC can produce. Full-scale is $0dB$ and $-20dB$ is one tenth of the full scale voltage which, when applied to your headphones or speakers, will produce 1/100 of the energy that full scale would produce.

4.2.2 Dynamic Range

This parameter attempts to capture the ratio of the loudest signal to the quietest signal. The loudest signal is limited by the electronics to a certain voltage, it may be as much as 4Vrms. The quietest signal is limited by noise within the electronics and may be as little 1μ Vrms. Dynamic Range is commonly specified by the acronym "DNR".

Dynamic Range is a measure of the signal to noise that you could, in principle, if you had perfect hearing, hear in the presence of music. If the brain succeeds in listening to the "gaps" (to the silence between the music) this is the noise level it would hear.

There is a "trick" that certain audio devices employ to reduce the noise and hence improve the dynamic range: the electronic system monitors the signal level, and when it detects silence, it shuts down certain noise making elements in the system and a technical measurement will show a very low noise. However, this "muting" of components is judged to be an unfair comparison since the silence may be "in the background, behind" a musical element in the program. If the electronic system was unable to enter this "shutdown" mode, what then would be the dynamic range? Surely this more accurately represents the listeners perception.

Consequently, in order to prevent a spurious shutdown in those devices (which are admittedly pretty rare these-days) the DNR test is done in the presence of a small ($-60dB$) tone at $1kHz$. The measurement equipment removes this small tone and adds up all the other energy in the signal and reports this sum, in ratio to the 0dB signal, as the Dynamic Range.

4.2.3 Signal to Noise Ratio

Signal to Noise is something of an abstraction: it is not necessarily what you can hear in any music program. It is simply the degree to which the system is silent when no music is playing.

This parameter attempts to capture the ratio of the loudest signal to the quietest signal in a similar way to the Dynamic Range measurement. But unlike DNR, Signal to Noise is assessed by simply measuring the output with no signal applied and expressing this as a ratio to a reference level – in the case of a DAC, the reference level is full-scale. Signal to noise is commonly specified by the acronym "SNR".

Note that if the device is employing some sort of muting, this may activate and make the SNR appear very high. It should be said that the muting trick is not entirely without merit, since it will reduce any hiss from speaker when the volume level is loud and the music (vinyl, tape or CD) has finished playing.

Given this description, you may well be surprised to see that the Invicta has an SNR that slightly exceeds its DNR: does this imply it uses a muting trick to reduce noise? The answer most assuredly is no - no such muting trick is employed. The Invicta and other high performance audio devices have a very high dynamic range and that dynamic range is limited by a phenomenon that causes the noise floor (as measured with no signal) to rise slightly in the presence of a signal.

This phenomenon comes from two sources. The first is the noise shaping loop of the ESS Sabre DAC itself. It, like all noise shaping loops, is a strongly non-linear device (non-linear in the mathematical sense, bordering on the chaotic) and the noise characteristic differs very slightly with signal level: the noise is slightly lower from the ESS Sabre DAC when it has no signal to process. This reduced noise is accurately delivered to the XLR connectors on the back of the Invicta because we take great care to introduce the absolute minimum of additional noise in the signal path. Our signal path noise is so low that the Sabre DAC noise change is visible in the specifications.

The Invicta does not use any tricks to artificially improve SNR.

Another source of noise in the presence of the signal only becomes apparent at very low noise levels and that is due to the finite gain of the op-amps used in the signal path. As the op-amp is forced to deviate from equilibrium to drive its own internal dominant pole capacitor (and perhaps the output load), its own noise level can rise. However, we see no evidence of this second source of noise in the Invicta[4].

[4]Probably because the ADI797 op-amp used in the signal path has a DC bootstrapped compensation capacitor (and it has an additional AC-coupled feedback

4.2.4 Frequency Response and Slew Rate

To what degree can the sound pressure waves that we hear be translated into electronic form and reproduced? To answer this question in its most basic sense requires us to understand the limitation of any system that attempts to process the analog of a "real-world" parameter in some other domain. This rather grand-sounding assertion is not difficult to understand if we break it down into its component parts. An example will help.

Frequency response and Slew rate are explained by analogy to a simple system in this section.

Suppose (prior to our high-tech world where we may make a phone call) that we want to know the temperature on an island a few miles from our research base on the mainland. To do this, we employ an attendant on the island and give him a flag that he may set to a certain height on a flag-pole that we can just see from the base. We tell our attendant that the lowest point on the flag pole is to represent zero degrees (0°C) and the highest point is to represent one-hundred degrees (100°C).

Each day he sets the flag to the appropriate height and we, being able to see it from our base, can then know the temperature on the island. The height of the flag has become analogous to the temperature: the height is an analog of the temperature. It is the height we see, not the temperature itself, but since the height is analogous to the temperature, we can note the temperature on the island.

Where does this system of making an analog of the "real-world" parameter (temperature) in the other domain (height of the flag) fail, and what artifacts is it prone to experience?

It should be obvious where the first problem may lie: what if the temperature goes to -10°C? This is beyond the range of the analogous quantity and the system fails. Similarly above 100°C. This primary failure mode contributes to limited dynamic range. But another potential failure relates to how quickly the flag can be raised or lowered. It should be no problem to get a temperature reading each day: the attendant can easily set the flag once each day, but what if we need to know the temperature every ten seconds?

Asking for the temperature every ten seconds causes a problem because it is difficult for the attendant to raise and lower the flag so frequently. Hence there is a frequency limitation in this analogous system - it

path) that vastly reduces the deviation from equilibrium in the input pair of JFETS.

cannot communicate changes in the temperature that occur faster than the attendant can raise and lower the flag. Any variation occurring on a timescale less than about ten seconds is not capable of representation in the flag height system due to a bandwidth limitation.

Analog systems exhibit a non-linear phenomenon called Slew Rate limiting. It can appear as an amplitude dependent frequency response and the human ear may be able to detect this imperfection.

Note that this has nothing to do with the system being digital: this is a analog system we are discussing here. There is no encoding of the height into a number, no discrete time at which the attendant can operate—in principle he can continuously adjust the height of the flag to any arbitrary accuracy. This is an analog system, but clearly it has a finite bandwidth.

But that is not all: what exactly prevents the attendant being able to change the flag every ten seconds? We can imagine that it is due to the need to pull the rope as quickly as possible from one height to the next, and no matter how he tries, that rope can only move through his hands at a certain speed. This gives rise to an interesting phenomenon called "slew rate limitation" that makes frequency response (in an analog system) a non-linear parameter. If asked how quickly he can signal a change in temperature the complete answer is this: if required to move just a few centimeters, as will be the case if the temperature changes a little, he can probably move the flag as fast as once every two seconds. But, if asked to move the flag over five meters or so, much further than before, he can change the flag only every ten seconds.

The frequency response in therefore non-linear: it depends on how large a change we are considering. In the realm of electronics there is a strong tendency when asked, "What is the bandwidth of your system?", to answer with what is more accurately called the small-signal bandwidth—that bandwidth where the flag need move only a little in our analogy. Any deviation from this bandwidth due to large signals is referred to as "slew-rate limitation"[5].

These two parameters, bandwidth and slew rate, are available for all op-amps produced today and the op-amp manufacturer expects the engineer to realize that the op-amp will not in fact show the bandwidth

[5]In fact, the situation is even a little more abstracted than this suggests. The small signal bandwidth is that bandwidth that would be present with an infinitesimally small signal—a mathematical abstraction due to the CAD (Computer Aided Design) tools used when designing a circuit. SPICE, the common design tool, performs AC analysis using a linearized model. True bandwidth analysis must use a less commonly available, but more advanced, method called Periodic Steady State (PSS) analysis.

specified in the presence of large signals where slew-rate limitation is dominant.

In audio equipment, if an input signal of fixed amplitude is swept from 10Hz to 50kHz and the output signal amplitude measured vs. this input frequency, then surely we have captured the frequency response of our audio device. Indeed we have, and it is likely that the bandwidth was around 20kHz. But we may ask this: when the bandwidth started to roll-off, was this due to the generation of harmonics at 40kHz, 60kHz etc. (the multiples of the input signal) or did the output signal reduce in amplitude without generating harmonics?

Slew Rate induced distortion may not show up in the output of the system as distortion, but it does begin to roll-off the in-band fundamental. Consequently, it makes a non-linear (amplitude dependent) frequency response.

If it was the former case, that is, if harmonics were generated, then we are likely suffering not from limited bandwidth per-se, but from slew rate limitation. In this case it is almost certain that the bandwidth is not independent of the signal amplitude: it will appear higher for a lower input signal[6].

Consequently, two systems of equal bandwidth, at a given signal level, may differ in detail. One may show a bandwidth limitation independent of amplitude and generate negligible distortion even as the bandwidth rolls off. The second system may show a variation of bandwidth with signal amplitude and generate harmonic content as the bandwidth rolls off. Given our increasing understanding of the human ear, it is almost certain that the audiophile will hear these systems that have the same technical specification of bandwidth, differently[7].

[6] Even this is not as clear as it may be and a full analysis must ask: "How did you measure the output signal when you were sweeping the input?". If the output signal amplitude was measured only at the input frequency, then the energy in the harmonics was not considered. We may well ask, "How does the ear perceive a finite bandwidth?". It may perceive an attenuation as the ratio of the output signal to the input signal at that given frequency, but it may perceive attenuation as the ratio of total energy output (including harmonics) to the input energy.

[7] Again, the knowledgeable reader will question this in at least two aspects. Firstly, surely this analysis is flawed since the system with a bandwidth dependent on amplitude will show a higher distortion and therefore a true A-B comparison of this phenomenon is not possible: the posited systems must differ in distortion specification and hence how can they be a fair comparison for the bandwidth issue? The answer is that in an audio system, distortion is specified up to a certain bandwidth. For example, the sum of the energy in harmonics up to 22kHz is compared to the fundamental and this is reported as the THD (Total Harmonic Distortion). Despite the name, this is only the "total" up to a given bandwidth, not the true total. Also, commonly, the THD test is done at 1kHz input signal. In a digital system, THD can be the true total due to the phenomenon of aliasing, but in an analog system, those high harmonics can go unaccounted for in THD. And, for this particular case of lim-

4.3 Fourier Analysis

Fourier Analysis is the use of a procedure that takes a signal expressed in time and transforms it to one expressed in frequency. This "Fourier Transform" is but one of a class of similar mathematical transforms that can change the representation of something reversibly from one form to another. Fourier transforms are common in audio applications. We may ask about the amplitude of our music, its start and stop time and so forth. These are all parameters of its expression in the time domain. It is interesting also to ask about its lowest frequency component, its highest frequency component and so forth; these are parameters of its expression in the frequency domain. In other words, of its Fourier transform.

This section describes a simple analogy to Fourier Analysis.

Fourier transforms may be appreciated by a simple analogy. Imagine that the consumption of electricity in New York was recorded every few minutes by the utility company for a period of some years. This electricity consumption record could be analyzed and a surprising amount could be learned. For example, we could ask, "What is the variation of electricity consumption in each day?". The daily variation would be significant as lights come on in the evening and meals are cooked. We could ask, "What is the variation of electricity consumption per week?". The weekly variation would show the difference of weekend consumption. Similarly, the yearly variation would show the temperature since heaters are on more in the winter. It is possible that evidence for global climate change would be present in the year to year variation. This method of asking, "What component in this data changes every day/week/year?", is Fourier analysis: the representation of the data as the amplitude of cycles of various lengths.

It is well known (and has been since the early 19th century) how to find the components of any frequency in any data set, but a particularly efficient means to rapidly search for *all* frequency components *at the same time*, did not come into general use until the mid 1960's and the

ited bandwidth, the harmonics are always high, since the bandwidth is presumably somewhere near 20kHz at least. Secondly, what meaning do higher harmonics above the bandwidth have—the bandwidth is limited (that's the whole point) and this removes them does it not? Actually, not necessarily. It depends on where in the signal chain the slew-rate limited bandwidth is occurring. It can be that the output stages are not slew-rate limiting, but the earlier stages are. In this case the harmonics of the slew rate limiting element are passed to the headphones or speakers, and despite being higher frequency, we now know the brain perceives them at some level.

advent of digital computers[8].

Certain parameters, frequency response is the obvious one, are clearly descriptions of the frequency domain representation of a signal, but others, such as distortion, are best appreciated in the frequency domain as well[9]. Distortion in the time domain is a deviation from the ideal amplitude that varies with the amplitude itself. Distortion in the frequency domain is conveniently collected into multiples (harmonics) of the fundamental signal and so is easy to appreciate, visualize and compare.

Analysis in the Frequency Domain is essential to uncover artifact and quality issues in an Audio system...

Having explained the significance of Fourier Analysis, it is probably a surprise to realize that the graphs that we see (including the ones presented in this document) are not the Fourier Transforms of the signal! They cannot be, because in the frequency domain a signal is not a real quantity: it becomes a complex quantity (in the mathematical sense) having both phase and amplitude. The graphs we produce on the measurement machines, and print in documents such as this one, are just the amplitude component of the complex quantity at that frequency.

The absence of phase information in the plots we see makes it impossible to correlate the perceived clarity of the music, its tone, precision and "warmth" with the graphs. One system that may sound pleasing to the ear may have a certain pattern of harmonic content (as from a tube amplifier) identical to another system, but the second system may sound bad. This is because the frequency plots do not show the relative phases of the harmonic content.

... but bear in mind that a Frequency Domain plot is not the whole story: no phase information is present.

The situation is even further compounded by the fact that the phase may vary non-linearly with amplitude of the signal, and it may not even be a "static" quantity: it conceivably may vary with time[10] which

[8]This "invention" of what is called the Fast Fourier Transform (or just FFT) has occurred at least two times in recent years and neither was actually new. It is remarkable to note that FFT, as it is called today, was known to Gauss (a brilliant German mathematician 1777—1855) in 1805.

[9]Noise is interesting since it can be appreciated in both the time domain and frequency domain in just about the same way: as a random deviation around the zero point.

[10]This suggestion, that the phase of a given harmonic may vary with time, may sound unlikely, but a pleasing Guitar amplifier exhibits a "collapse" (voltage drop) of the tube based rectifier circuit when a large power output is called for. The power supply recovers over many milliseconds and during that time the tube amplifier is undoubtedly experiencing a changing frequency response which will translate into a time dependent phase shift.

the brain is likely to interpret as a modulation of the distance to the sound source.

Instrumentation based on high performance ADCs gathering a record of data can preserve both phase and amplitude and further can capture the time dependent aspects. Instrumentation based on a swept filter and RMS meter cannot capture phase or time variation.

4.4 Switching and Linear Power Supplies

Switching power supplies are low cost and efficient – but they are virtually impossible to use in noise sensitive applications because they radiate electro-magnetic energy with high slew rate and high frequency.

The cost and weight of "magnetics", that is inductors and transformers, can be relatively high, particularly when the frequency at which they operate must be low. Classically, a power supply operating at 50Hz (the mains supply in Europe for example) needs to be relatively large, heavy and expensive. How much better power supplies could be if they could run at a rate of 100's of kHz. Indeed they would be better: the faster that the circuit operates the less the size, the lower the cost of the magnetics that it needs. Hence the commercial pressure to develop a new kind of power supply: the switching power supply.

All new things win in the market place when they have just sufficient functionality, and crucially, lower price. But if they can add a benefit at the same time, then their adoption will be rapid. A switching power supply manages all three and its additional functionality is the ability to operate from 115V to 250V without user intervention.

In consumer electronics the art and manufacture of switching power supplies is so advanced that rarely does a manufacturer bother to design a new one. Stock switching power supplies in consumer electronics are today a commodity item provided by the subcontractor: you may have your product assembled in the Far East and the contractor will provide his own low cost power supply for your product.

Switching power supplies have enabled low cost, good quality consumer audio and, together with Class-D chips, a 7.1 surround sound home theater may be purchased for less than the price of the early DVD players.

Why, then, not use a Switching Power Supply every time? Sadly, the very things that enable the switching supply to work, namely high operating frequency, good efficiency and low cost magnetics, prevent its

use in high precision, very low noise circuits. The high frequency aspect radiates electro-magnetic noise in the vicinity and the high efficiency means fast slew rates and high currents, which again contributes to high emissions. The unit may be enclosed in a metal box, the IO may be inductively snubbed, but experience shows that despite great efforts, the noise will somehow break out and be detectable in the low noise circuity.

For all these reasons, Resonessence uses conventional, low operating frequency (mains frequency) magnetics. Still decoupled with series inductors, and smoothed with vastly over specified capacitors[11], the linear power supply feeds high performance linear regulators to distribute power[12].

Invicta uses enhanced linear power regulation for the lower noise possible.

4.5 An Invicta from 1935

It seems that the name "Invicta" has been used for an electronic audio device for many years. In approximately 1935 the name was used on a tube (valve) radio, notable because of its use of eight preset stations.

"Invicta" has been used as the name of electronic audio devices before: it is visible on this 1935 tube radio.

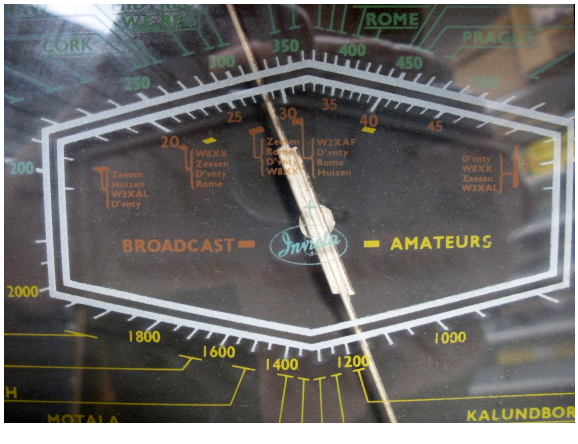

Figure 4.1: Tuning window of the "Invicta" 1935 tube radio

This example of the Invicta is from the Amgueddfa Ceredigion Museum

[11] Because electrolytic capacitors are the weak spot in long term life

[12] But even these high performance regulators are not sufficient: they are supported with discrete and additionally filtered devices that handle the primary current flow. The off-the-shelf high performance regulator chip is used for its internal reference, current limit and closed loop controller only.

in Aberystwyth, Wales. Resonessence Labs express their thanks to J J Danks (Jez) of the Aberystwyth Museum for kindly providing access and photographs of the 1935 Invicta[13].

Electronic components have changed beyond all recognition since 1935 but, even today, the circuits designed by the pioneers of radio are still used.

Tube radios, such as the Invicta pictured here, used amplifier configurations in tube technology that are in use as transistor configurations today.

Figure 4.2: The Invicta tube radio from 1935

[13]Resonessence Labs has no connection to the manufacturer of this radio, but we were pleased to see that the name "Invicta" was associated with an electronic device as long ago as 1935.

Chapter 5

System Components

5.1 Inside the Invicta

Digital and Analog sections inside the Invicta are carefully separated. Metal walls prevent electro-magnetic coupling.

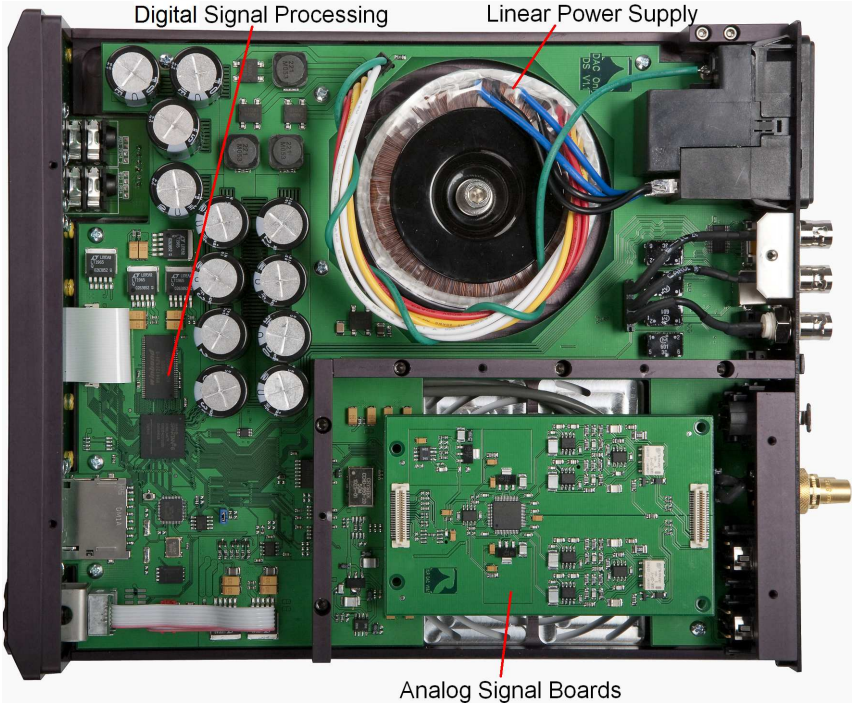

Figure 5.1: Invicta Internal Components

- Invicta uses a linear power supply and a toroidal transformer. This helps to reduce induced noise and ensures that any high current paths are, at most, moving at the mains supply frequency. The toroidal transformer helps contain any electro-magnetic fields generated. All the capacitors are over-specified in order to increase lifetime and each regulator chip is noise-reduced by addition of a discrete buffering stage such that power supply currents actually flow in well-controlled discrete devices.

The analog signal paths are separated onto small, individual boards. Ground currents for each board (XLR, Headphone etc.) are managed separately.

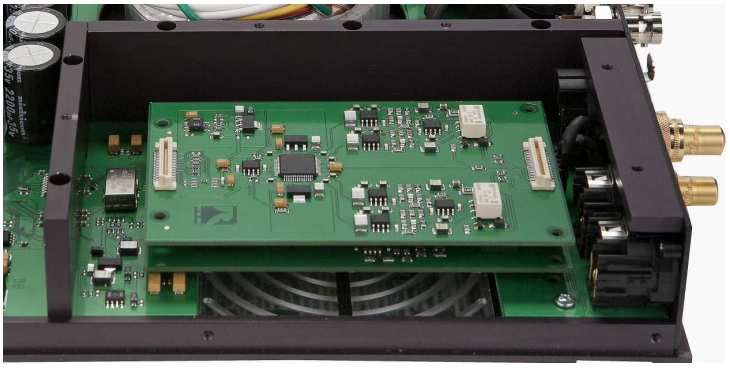

Figure 5.2: Analog boards

- To maintain the performance of the ESS Sabre DAC, its power supply is generated locally on a separate board. This is key to management of ground currents and power supply induced noise. On that same board, amplification and signal conditioning circuits associated with the particular function (XLR or Headphone) are assembled in a crafted current environment. Supply, ground and signal currents are managed separately – a ground plane is not used, but a flooded area on the board is present to define boundary conditions for any electro-magnetic induction. To a first order, no current flows in this flooded area[1].

- The digital engines of the Invicta are a series of MicroBlaze[2] soft processor cores instantiated as needed in the Invicta FPGA. The cores run independently and microcoded routines handle all user

[1] Experience teaches that the ground plane is not used to carry any DC currents: these DC currents are sufficient to induce ground coupling at measurable levels. Rather, it is better to provide discrete traces on the board to mange current flow precisely and leave the "ground plane" as a magnetic and electric field shield.

[2] "MicroBlaze" is the name for the Xilinx soft processor core created within the multi-million gate FPGA of the Invicta.

IO and rear panel interfaces. Critical audio paths in the digital engine are "hardcoded" meaning that the bit widths, truncation errors and so forth are all managed.

The clock (timing source) is critical. Although the Sabre DAC has on-chip re-synchronization to its master clock, higher performance can be achieved by controlling the relative phase of the system and the DAC clocks.

- Invicta uses an OLED (Organic Light Emitting Diode) display to reduce noise. The more common VFDs (Vacuum Fluorescent Display) are bright and clear, but require a relatively high drive voltage and switching speed which tend to radiate noise. OLED displays operate with low level signals and experience shows that noise can be managed more effectively with this type of display.

- The master clock of the Invicta is a precision, ultra-low phase noise oscillator. The precise phase and frequency relationships of these clocks to the audio stream is crafted to best exploit the Asynchronous Sample Rate Converter in the ESS DAC. The resulting audio signal actually exceeds the performance of the industry standard measurement machines as verified by an ultra-high performance ADC. This graph shows non-A-weighted (wide-open bandwidth to 96kHz) THD+N of the ES9018 in the Invicta vs. the outstanding AP2722 signal generator.

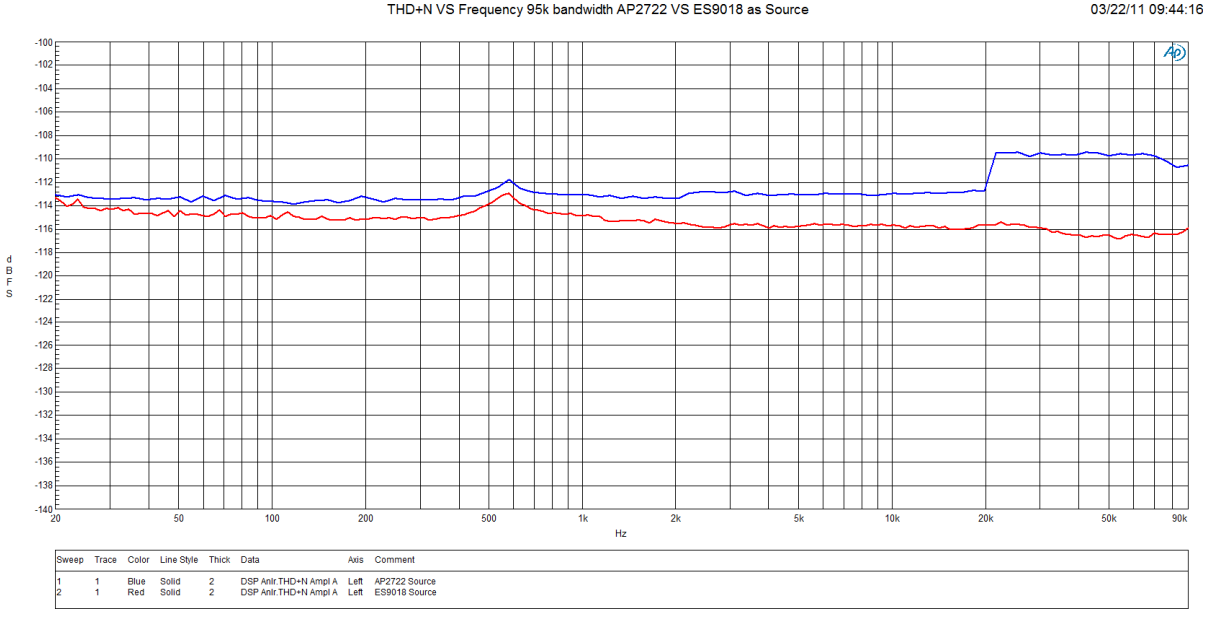

Figure 5.3: Invicta ES9018 compared to Audio Precision AP2722

5.2 Notes on Components

Passive components can easily be overlooked. They seem insignificant compared to the complexity and time taken to construct the DAC, USB or Micro-Controller but, it would be a mistake to underestimate the impact that a poor choice in passive components can have.

Modern specialist Surface Mount Components have the performance needed for Invicta and they have less inductance, and hence, lower noise than through-hole components.

The first high quality audio products were characterized by through-hole high performance passives: typically the German WIMA capacitors were used and they did indeed justify the excellent reputation that they acquired. Non-inductive resistors were a necessity and the skilled audio engineer could pass judgment on a design before even turning it on – he simply visually inspected the board and saw if the manufacturer knew what he was doing[3].

The Invicta uses surface mount components. These are not common surface mount, but specially selected low inductance and high linearity versions. For example, a common surface mount capacitor would ruin the THD, but the special high linearity types do have the performance needed.

[3]Even the DAC's had a certain look to them: a 28 pin TSSOP or similar became associated with high quality. The outstanding ESS Sabre DAC did not fit this pre-conception and it was initially dismissed as unlikely to succeed. That has now changed; the ESS Sabre DAC is the acknowledged leader in DAC technology.

Chapter 6

Specifications

These specifications are a summary of the data collected and presented in Chapter 3, the Technical Measurement section.

These specifications are a summary - you may refer to the detailed results in Chapter 3.

Figure 6.1: Invicta Dimensions

Specification	Value	Units	Notes
General			
Width	22.0 (8.66)	cm (in)	
Depth	28.2 (11.10)	cm (in)	
Height	5.0 (1.97)	cm (in)	Including feet
	4.4 (1.73)	cm (in)	Without feet
Weight	2.9 (6.4)	kg (lb)	
Operating Voltage	110 - 130	V AC	60Hz, 115v setting
	220 - 250	V AC	50Hz, 230v setting
Fuse Rating	1.0	Amp	Fast, 250V, 1.25in
Power Consumption	30	Watts	
SD Card File Format	WAV, AIFF		Up to 192kS/s, 24bit
XLR Output			
Amplitude	4.6 ±5%	Vrms	At $0dB$ volume
Impedance	156 ±1%	Ω	
Distortion + Noise	< -110	dB	$1kHz$
	< 0.00032	%	
Dynamic Range	> 125	dB	A-weighted
Frequency Response	−0.25	dB	at $20kHz$ $44.1kS/s$
	−1.0	dB	at $50kHz$ $192kS/s$
	−3.0	dB	at $80kHz$ $192kS/s$
RCA Output			
Amplitude	2.3 ±5%	Vrms	At $0dB$ volume
Impedance	78 ±1%	Ω	
Distortion + Noise	< -110	dB	$1kHz$
	< 0.00032	%	
Dynamic Range	> 120	dB	A-weighted
Frequency Response	−0.25	dB	at $20kHz$ $44.1kS/s$
	−1.0	dB	at $50kHz$ $192kS/s$
	−3.0	dB	at $80kHz$ $192kS/s$
Headphone Output			
Amplitude	5.1 ±5%	Vrms	At $0dB$ volume
Impedance	1.0 ±1%	Ω	
Distortion + Noise	< -100	dB	$1kHz$ unweighted into 60Ω
	< 0.001	%	
Dynamic Range	> 118	dB	A-weighted
Frequency Response	−0.25	dB	at $20kHz$ $44.1kS/s$
	−1.0	dB	at $50kHz$ $192kS/s$
	−3.0	dB	at $80kHz$ $192kS/s$

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